

Study Group 14

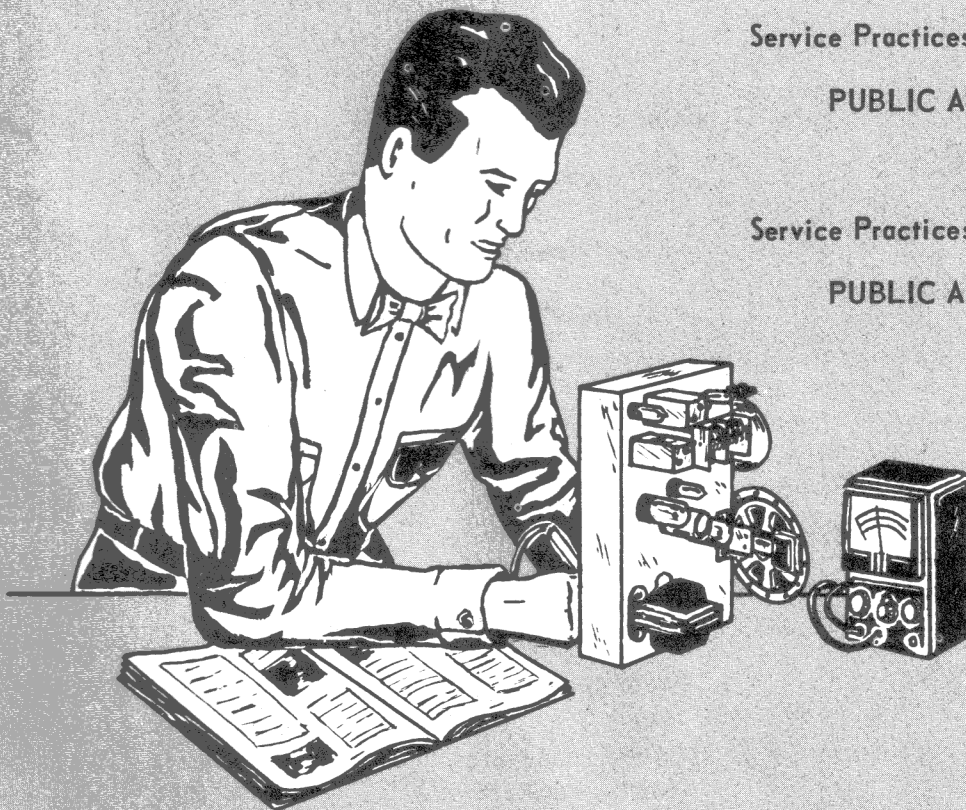
ELECTRONIC FUNDAMENTALS

Service Practices 27:

PUBLIC ADDRESS SYSTEMS I

Service Practices 28:

PUBLIC ADDRESS SYSTEMS II



RCA INSTITUTES, INC.

A SERVICE OF RADIO CORPORATION OF AMERICA

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ELECTRONIC FUNDAMENTALS

SERVICE PRACTICES 27

PUBLIC ADDRESS SYSTEMS I

- 27-1. A Simple PA Amplifier Circuit**
- 27-2. Mixer Circuits**
- 27-3. Power Amplifiers**
- 27-4. Output Circuits**



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Service Practices 27

INTRODUCTION

The function of a public address (PA) system is to raise the volume of a speaking voice, music, or some other sound for the benefit of people who otherwise could not hear the sound. In this booklet, we take up the electronic circuits employed in public-address equipment. In the next booklet we take up use of the equipment.

Public-address systems center around an audio amplifier. The amplifier works on the same principle as the audio amplifiers that you have already studied. But you are accustomed to think of an audio amplifier as one of the several circuits that make up a radio receiver. In public address systems, the set up is somewhat different. Audio is the only kind of signal that is processed. There is no r-f, no i-f, and no oscillator signal. Therefore, only audio circuits are contained in public-address equipment.

The simplest PA amplifiers have circuits much like the audio circuits of radio sets. Sometimes many loudspeakers must be attached to PA equipment. PA equipment must be connectable to various sources of audio signal. Furthermore, sometimes PA equipment must be able to handle several sound inputs at once. More complicated circuits than radio audio circuits are used when special needs must be filled.

If an amplifier is to drive many loudspeakers, it must be able to deliver more power than an amplifier driving only one speaker, and special output circuits must be employed. If audio input from various sources is to be handled, special amplifier input circuits must be employed. Some input circuits handle several sound inputs at once.

In this booklet, we start with a simple radio-like circuit and progress to more elaborate circuits. You will learn new features of the more elaborate audio circuits, but you will not have to relearn the basic principles of audio amplifier operation which, you will find, apply even to the elaborate circuits.

27-1. A SIMPLE PA AMPLIFIER CIRCUIT

Over-all Discussion. If you examine the schematic of a simple PA amplifier circuit in Fig. 27-1, you will probably recognize that it is a three-stage audio amplifier. The first stage, labeled *input section*, is a voltage-amplifier stage that you have not come across in radio receivers. Its purpose is to amplify the low input to the PA system enough to drive the middle stage.

Probably you recognize the middle stage, called the *main amplifier*, as a circuit much like the audio-amplifier circuits that you have studied. V_2 , a voltage amplifier pentode, drives V_3 , a power amplifier pentode.

However, you may not be familiar with the application of *negative feedback* (also called *inverse feedback*), which is used in the main amplifier section even though you have already learned what the term means. Feedback from the plate of V_3 is returned to the plate of V_2 through the feedback resistor shown in the schematic.

Negative feedback improves frequency response, reduces distortion, and maintains the output voltage of the amplifier at a constant level for a wide range of load impedances provided that the input to the amplifier is constant. The feedback does not reduce the maximum possible power output of

the power tube, but it makes necessary an increase in the driving signal applied to the grid of the power tube.

The third stage, labeled *output section*, consists of the amplifier output and loud-speaker.

A manufacturer might list the specifications of the PA circuit as shown in Table A.

TABLE A

Maximum power output:	8 watts
Output impedance:	4, 8, or 16 ohms
Frequency response:	80 to 10,000 cps, down 3 db at 80 cps.
Inputs:	One high-impedance microphone and one high-impedance phonograph. The microphone input can be mixed with the phonograph input.
Gain:	The gain from the microphone input is 100 db. The gain from the phonograph input is 70 db.
Input impedances:	Microphone: 1/2 megohm Phonograph: 1/2 megohm

Details of Circuit. The circuit shown in Fig. 27-1 can be used with either a microphone or a phonograph. When used with a phonograph, the phono input voltage is high enough to drive the amplifier without preliminary amplification. However, when a microphone is used with the circuit, the input voltage from the microphone is not high enough to be sent to the main amplifier. The gain needed is provided by V_1 , the microphone preamplifier in the input section.

The isolating resistor isolates the internal impedance of the phonograph pickup from the output impedance of V_1 . The phono is permanently connected to the output of V_1 . Without an isolating resistor, the phono impedance would tend to short circuit the output of V_1 and, at the same time, the output impedance of V_1 would tend to short circuit the phono. The best way to visualize how the resistor performs its function is to look at it from two standpoints — from the standpoint of the phono and from the standpoint of V_1 . The phono sees the resistor in series with V_1 . Because of the resistor, the phono sees a larger total impedance than it would see if it were directly connected to

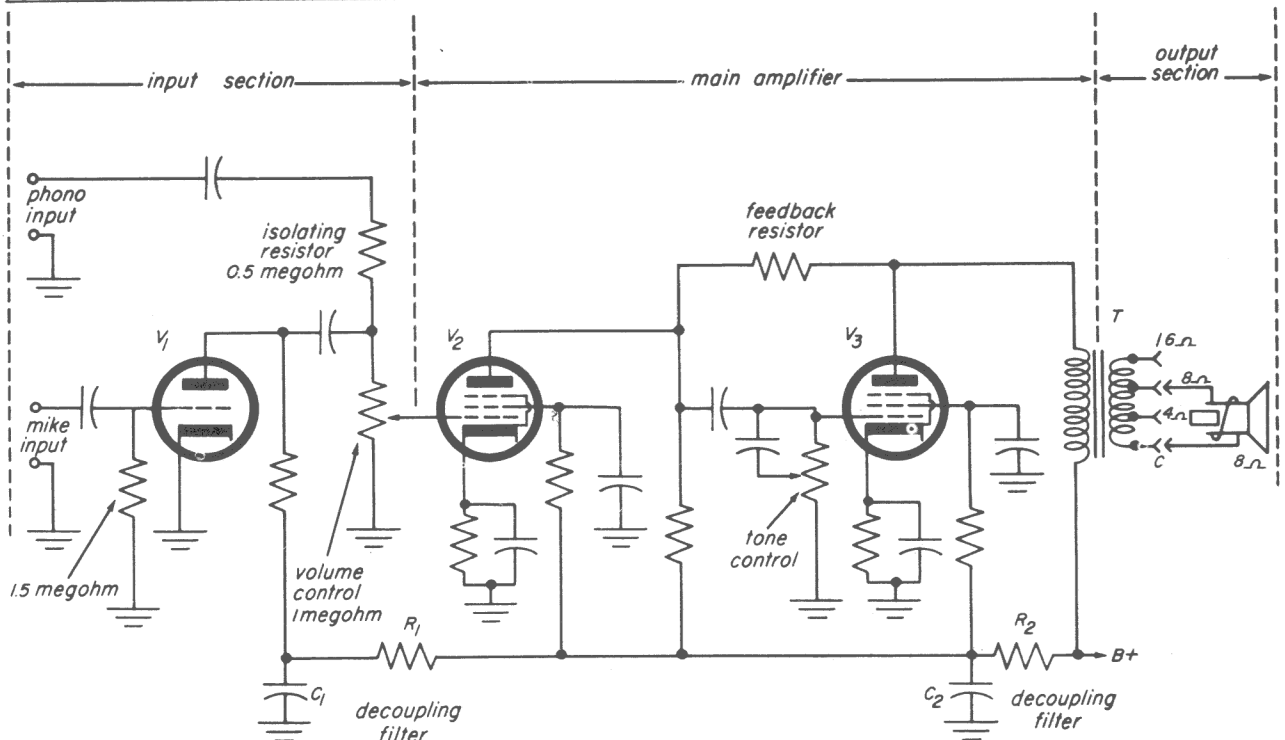


Fig. 27-1

V_1 . At the same time, V_1 sees the resistor in series with the phonograph.

With the phono input permanently connected to the output of V_1 , the phono-input signal can be mixed with the microphone input signal. When an amplifier stage is wired in this way, it is called a *mixer*. The input section of this audio amplifier is called a *microphone preamplifier and mixer* circuit. The permanent connection feature of mixer circuits is advantageous even when you don't want to mix the input signals; with it, you can have all inputs plugged in and ready to go at all times. No input selector switch need be provided. Some kind of mixer circuit is always employed in the input sections of PA amplifiers.

The input section drives V_2 . V_2 is a pentode instead of a triode because a pentode provides more gain than a triode. The large gain is necessary because V_3 requires a large driving signal.

If there were no phono input to the circuit shown, V_1 could have been a pentode and V_2 could have been a triode. The total gain would have been the same as in the arrangement shown in the figure, and the mike input would be amplified enough to drive V_3 . However, since there is a phono input that must be amplified enough to drive V_3 , and there is a loss of phono voltage across the isolating resistor, making V_2 a pentode provides the additional gain needed for the phono input.

Considered as a single unit, the input section and the main section make up a three-stage amplifier. All amplifiers that have three stages or more and that use all three stages to process the same signal, need decoupling filters in the B+ section. Let's consider the decoupling filters in the circuit of Fig. 27-1.

The circuit shown has two decoupling filters. One consists of C_1 and R_1 ; the other consists of C_2 and R_2 . These filter out all signal voltage on the B+ wiring, leaving only d-c voltage. If there were no decoupling filters, some of the signal at the plate of

V_3 , would be coupled by means of the B+ wiring back to the plates of V_2 and V_1 .

So far as the circuit of V_2 is concerned, the feedback would do no harm. The signal at the plate of V_2 is out of phase with the signal at the plate of V_3 . Without decoupling, the V_3 signal would cancel some of the V_2 signal. This is negative feedback, and negative feedback is deliberately applied to V_2 from V_3 by means of the feedback resistor. If we had only V_2 and V_3 to consider, we would have a two stage amplifier, so we wouldn't have to worry about decoupling. The phase relationship of signals at the plates of all two stage amplifiers are such that only negative feedback can take place via the B+ wiring. Most radio receivers have two-stage audio circuits with no decoupling.

However, if there were no decoupling filter in the circuit of Fig. 27-1, some of the voltage from the plate of V_3 would be fed back to the plate of V_1 as well as to the plate of V_2 . The signal voltage that is normally present at V_1 is in phase with the V_3 signal voltage. The two signals are in condition to reinforce each other. This is positive feedback or regeneration. It causes distortion and other bad effects. At worst, regeneration — the feedback of an in-phase signal — becomes oscillation. An oscillating amplifier generates an unwanted sound all by itself with no input signal applied. It generates a sustained tone. The amplifier becomes an oscillator. This seems logical when you remember from the oscillator lesson that regenerative feedback is a necessary feature of oscillator circuits.

RC decoupling filters work much like the LC filters in power-supply circuits. In the case of a power-supply filter (Fig. 27-2a), unwanted ripple voltage is filtered from B+. The ripple is at power frequencies (60 to 120 cps). Again, in the case of the decoupling filter (Fig. 27-2b) unwanted ripple voltage is filtered from B+. But here the ripple is at audio-signal frequency (80 to 10,000 cps for the circuit of Fig. 27-1). In the case of the power filter, unwanted ripple is blocked out

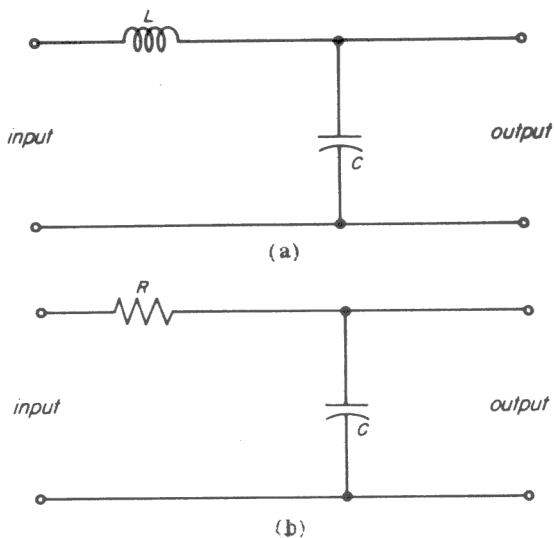


Fig. 27-2

of B+ by the choke coil and shorted out of B+ by the filter capacitor. In the case of the decoupling filters the unwanted ripple is blocked out of B+ by the resistor and shorted out of B+ by the capacitor.

In the output section of the circuit of Fig. 27-1, there are several taps on the secondary winding of the output transformer (T). These provide a way of matching speakers with different voice coil impedances to the output tube. The tap labeled C is the common tap to which one lead from the speaker load is always connected. Which of the other taps is used for the remaining lead from the speaker load depends upon the impedance of the speaker load that is connected. For example, a speaker with an 8 ohm voice coil should be connected to the tap labeled $8\ \Omega$ as shown in Fig. 27-1. When this is done, the transformer reflects the proper impedance to the plate of V_3 . The $8\ \Omega$ label can be understood to mean that *when an $8\ \Omega$ load is connected to this tap, the correct impedance will be reflected to the plate of the power output tube*. If you interpret all the tap labels in this way, it will be easy to remember the following rules governing use of the taps:

1. Normally, only one tap in addition to the common tap can be used. If a 4-ohm speaker is connected to the $4\ \Omega$ tap at the same time that an 8-ohm speaker is connected to the $8\ \Omega$ tap, two impedances will be reflected back to the output tube. The combined value

of the two reflected impedances cannot be correct for the tube, since the meaning of the tap labels tell you that one of the impedances alone reflects the correct impedance.

2. To connect more than one speaker, figure out the total impedance of the combined speakers and then connect the speaker combination to the tap that has a label corresponding to the total you figured out. For example, suppose you want to match to the amplifier a pair of 8-ohm speakers connected in parallel. You know that 8 ohms in parallel with 8 ohms makes a total of 4 ohms. So the pair of speakers combined in parallel should be connected to the tap labeled $4\ \Omega$. If the pair of speakers were combined in series, the total impedance of the combination would be 16 ohms and the series combination would have to be connected to the $16\ \Omega$ tap.

Now that you are familiar with the general principles of PA systems as they apply to a simple circuit, let's discuss more elaborate circuits.

27-2. MIXER CIRCUITS

Figure 27-3 is a schematic of only the input section of Fig. 27-1. This is a mixer circuit. It has certain features that are important to understand.

1. The circuit of Fig. 27-3 is designed to mix a signal of low-voltage level, the signal from the microphone — with a signal

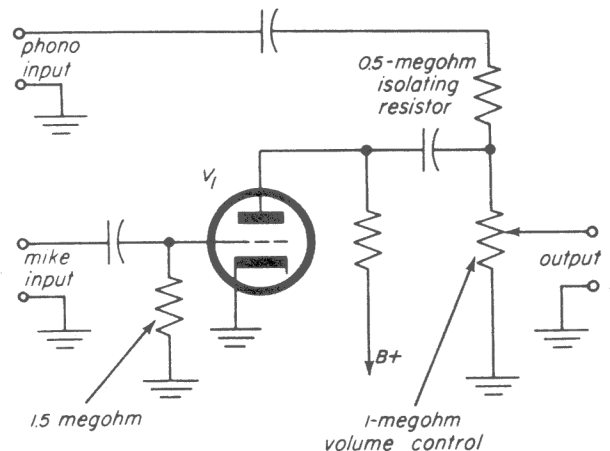


Fig. 27-3

of a higher voltage level — the signal from the phonograph.

2. The circuit is designed to receive both the high- and low voltage-level signals from high-impedance sources.

3. Gain for the two input signals cannot be controlled separately, but the amount of the combined signal fed out of the circuit can be controlled by means of the volume control.

Input Terminals. Because the PA circuit must amplify inputs from different sources, it has separate input terminals. First, let's consider the phono input terminals. We have already discussed briefly the fact that the input from a phono does not require amplification before it reaches the main amplifier section because the voltage from the phono is relatively high. Because the phono is a device with a high impedance, the phono input circuit also has a high impedance.

Although the input terminals where the phonograph can be attached are labeled phono terminals, you can attach other devices that have a high voltage level and a high impedance to these terminals. Some devices of this kind are:

1. Crystal phonograph pickups.
2. AM-FM tuners.
3. Variable reluctance phonograph pickup preamplifiers.
4. Tape recorders.
5. Microphone preamplifiers.
6. PA preamplifier/mixer units that are supplied separate from PA power amplifier units.

The other input terminals shown in the figure are labeled *mike input*. These terminals are provided for the connection of devices with a high impedance but a low output voltage level. (You have already learned that the low voltage level is increased by V_1 . About the only devices used with a PA system that have a high

output impedance but a low-level output voltage are microphones. The only microphones that do not come under this classification are low-impedance magnetic mikes. Since only microphones are connected to the terminals we are now discussing, these terminals are labeled *mike input*. This should cause you no confusion if you remember that the mike input is for all microphones except *low-impedance magnetic mikes*.

Low-impedance magnetic mikes have an output voltage that is too low to drive the voltage amplifier V_1 . Therefore, the voltage and the output impedance are stepped up by means of a transformer. Then the output of the transformer is applied to the mike input terminals. If this transformer is built into the PA system, the input to this transformer is labeled *low-impedance microphone input*.

Another type of input that cannot be applied directly to the high impedance mike input terminals shown in Fig. 27-3 is the voltage output of some kinds of magnetic phonograph pickups. Here the problem is different from the low impedance mike. The voltage doesn't need to be stepped up, because the voltage level from the pickup is high enough already. The impedance can be matched by substituting a resistor equal in value to the pickup impedance in place of the 1.5-megohm grid resistor in Fig. 27-3. Although it is best to have the low-impedance magnetic pickup connected to input terminals of the proper impedance, even this is not the major problem.

As you will learn in more detail later in this lesson, it is possible to connect a low-impedance device to an input of a higher impedance without too much loss of efficient reproduction. The main problem in the case of the pickup we are discussing is equalization. (We have discussed equalization in earlier Service Practices booklets). Equalization is usually provided by a separate preamplifier, and the output of this preamplifier is applied to the phono input of the PA circuit. The preamplifier output is at the appropriate voltage and impedance level.

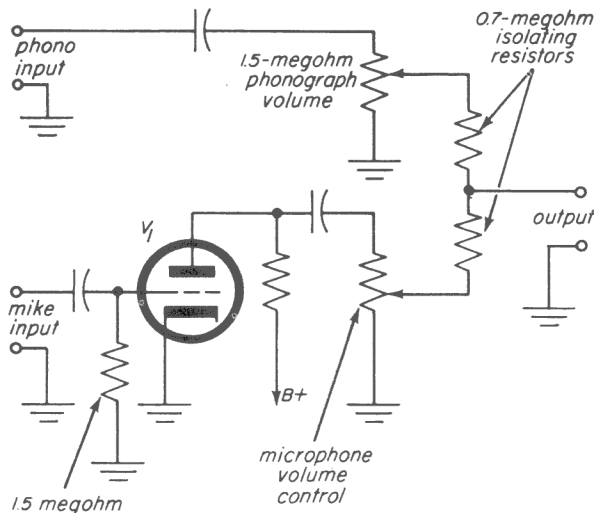
Impedance Levels and Voltage Levels.

From our discussion of the input terminals of the PA system, you can see that the main reason for the different inputs is that different devices that are used with the PA system have different impedances and different voltages. A device may have a high impedance and a high voltage as does a crystal phono pickup. It may have low impedance and a medium voltage, as some low-impedance magnetic pickups. It can have a high impedance but a low voltage, as do certain mikes. The main thing to remember at this point is that just because the voltage of a device is high, you cannot assume that its impedance is high.

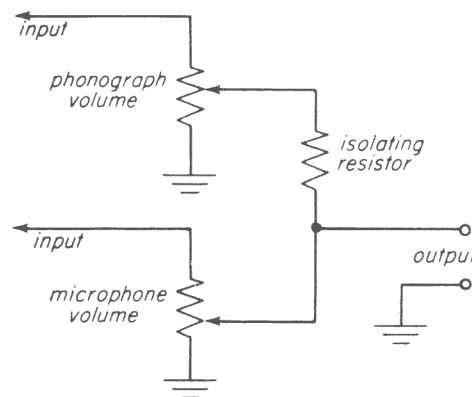
More Complex Mixer Circuits. Now that the features of the mixer circuit of Fig. 27-3 have been examined, we can consider some more elaborate mixers and you will understand their features.

To begin, let us modify the circuit of Fig. 27-3, so that the volume of the microphone and phonograph signals can be controlled separately. Figure 27-4 shows how this can be accomplished. The microphone volume control is in the same position in the circuit of Fig. 27-4a as the over-all volume control in Fig. 27-3. In Fig. 27-4a a volume control potentiometer has been added across the phono input terminals. The output signals from the two volume controls are brought together at the junction of two isolating resistors. It is not possible to employ a single isolating resistor as in Fig. 27-3. Look at Fig. 27-4b. This is not a usable circuit. You can see that where one isolating resistor is used, the phono signal is shorted out when the mike volume is turned to zero even though the phono volume is set high.

There is some interaction between the volume controls of Fig. 27-4a even though two isolating resistors are employed. Interaction in this case means that the phono circuit is affected by the operation of the microphone volume control and the microphone is affected by the operation of the phono volume control. An explanation of the interaction is as follows. The phono *sees*



(a)



(b)

Fig. 27-4

various impedances at various settings of the microphone volume control. The output of the phono varies with the impedance it *sees*. The same can be said of the preamplifier (V_1). It is because two potentiometers are connected together that interaction occurs. No interaction occurs in the circuit of Fig. 27-3 where only one control is employed.

The interaction (Fig. 27-4a) is slight and the circuit is practical but there is a way to completely avoid interaction. We must explain why this way is not followed. As you may have figured out, separate control of microphone and phono volume can be accomplished with only one control in the output part of the circuit as in Fig. 27-3 by substituting a volume control for the 1.5 megohm fixed-value grid resistor of V_1 . The arrange-

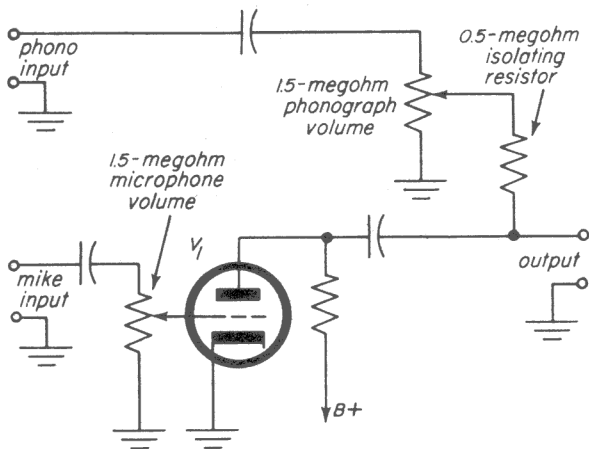
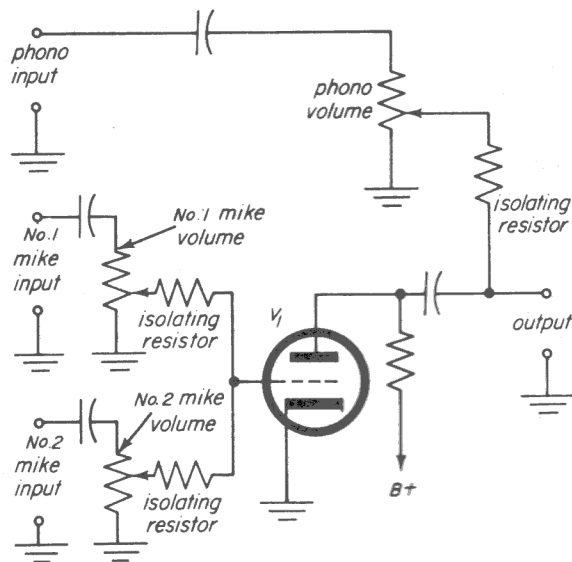


Fig. 27-5

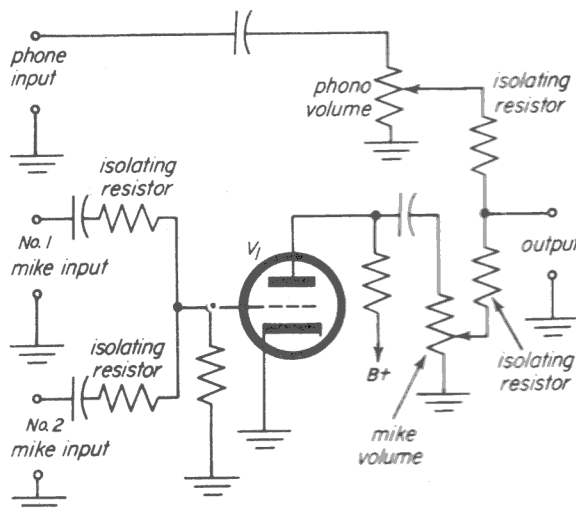
ment is shown in Fig. 27-5. It avoids interaction, but it is not used in practical circuits because it makes for a noisy control. The noise that is always generated when the moving contact slides across the volume-control resistor gets amplified too much when the volume control is located at a microphone preamplifier input. On the other hand, positioning a volume control before a preamplifier in the high-level phono-input channel does no harm because the signal at that point is strong enough to drown out the noise. A preamplifier is sometimes found in the high-level phono-input channel. We will show this circuit presently. First, let us consider the addition of a second microphone channel.

Let us devise a mixer circuit with two microphone-input channels and one phono input channel and with a separate volume control for each channel. The circuit of Fig. 27-6a is one possibility. This circuit is not satisfactory because the volume controls are positioned at the microphone input and they will be noisy. The circuit of Fig. 27-6b does not have controls at the microphone input. Therefore, it does not present a noise problem. However, it makes no provision for separate volume control of each of the two microphone signals. Once the two signals combine at the input of V_1 it is not possible to separate them at the output of V_1 . So the signals cannot be controlled separately at the output of V_1 . The circuit is not satisfactory for our purpose but it is satisfactory where separate microphone volume control is not wanted.

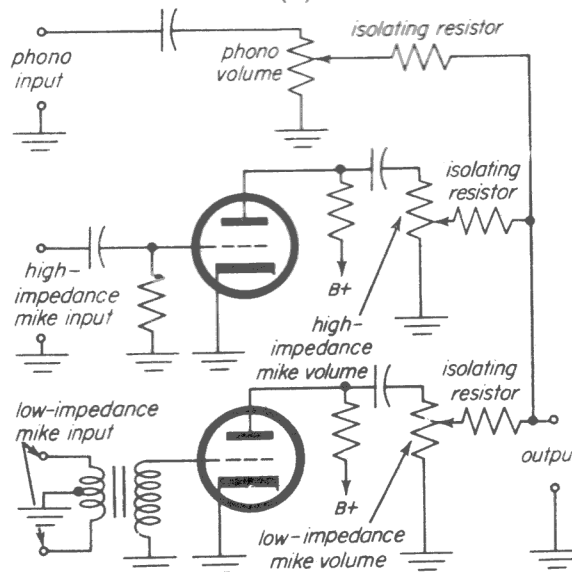
A good way to accomplish our purpose is to employ two electron tubes — one for each



(a)



(b)



(c)

Fig. 27-6

microphone — as shown in Fig. 27-6c. We have arranged one of the microphone channels for low-impedance microphone input to illustrate how this is accomplished. Alternately, both microphone channels can be either high-impedance or low-impedance channels.

The circuit of a high-level phono preamplifier is illustrated in Fig. 27-7. The microphone part of the Fig. 27-6c circuit is shown combined with a circuit having a phono preamplifier. Compare the microphone part of the circuit with the phono part. Volume control for the phono channels is located at the input to the tubes, that for the microphones is located at the output of the tubes. Because the phono signals are separately controlled at the input of the tubes, it is not necessary to keep the phono signals separated at the output of the tubes. There-

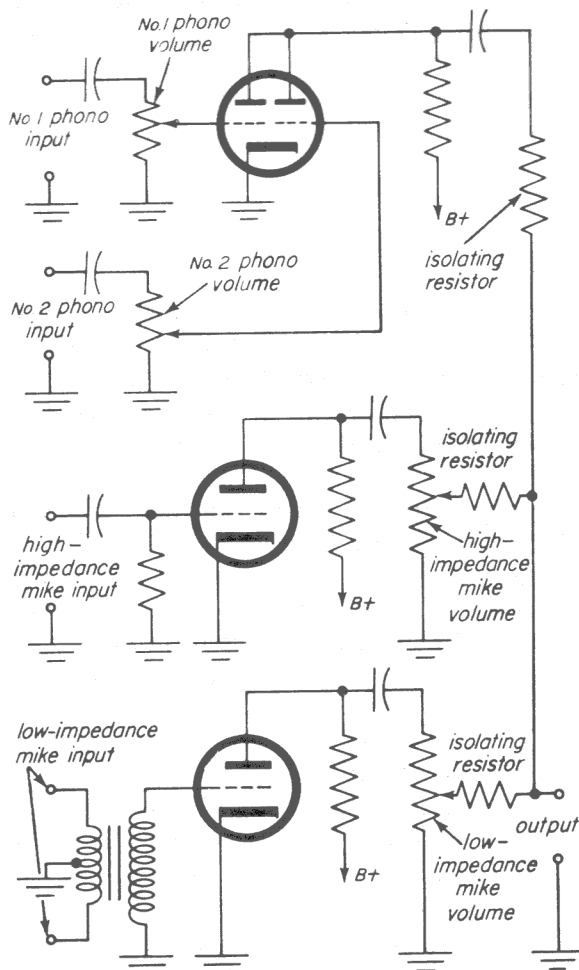


Fig. 27-7

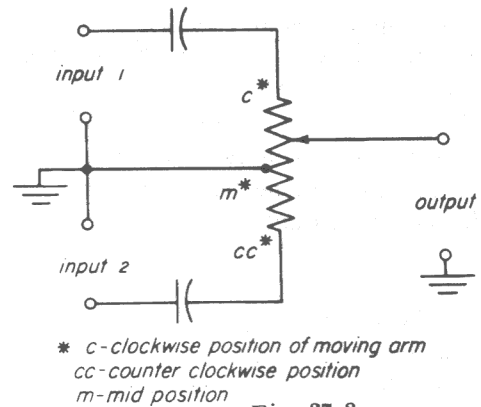


Fig. 27-8

fore, a single plate-load resistor can be used for the two phono preamplifier tubes.

Equipment with the circuit of Fig. 27-7 can be used to cause sounds from up to four sources to be reproduced all at once in a loudspeaker system. The loudness of one or more of the sounds can be changed at will. One sound can be faded out completely, then another faded in, etc.

Fader Circuits. Sometimes the main function of a system of PA controls is to fade one sound out gradually and then gradually fade another sound in. To do this with the circuit of Fig. 27-7, you turn the volume control of one channel down slowly; then you slowly turn the volume control of another channel up. A circuit that permits fading to be carried out with a single control is called a *fader circuit* (Fig. 27-8). The control works as follows. Assume the control is set fully clockwise and the sound from one channel is heard, full blast, from a loud speaker. You turn the control counterclockwise. As you do so, the loudness of the sound decreases. Midway between the full clockwise and the full counterclockwise settings of the control, the sound dies out completely. As you continue past the mid-point in the counterclockwise direction, the sound from another channel commences. It gets louder as you continue further counterclockwise, and is loudest at the full counterclockwise settings of the control.

The circuit, shown in Fig. 27-8, is like a mixer. It provides for two separate inputs and a single output. Unlike the mixer, it has no provision for the input signals to appear together at the output terminals. One or the

other of the input signals appears at the output terminals, depending upon the setting of the control. Fader controls make for a smooth changeover from one channel to another. Where the sounds from the two channels never have to be heard together, a single fader control is better than two volume controls.

Although there are many more mixer circuits, enough have been considered to illustrate the idea of input mixing as it is found in PA equipment. Now, let us take up the main amplifier sections in which audio power is developed.

27.3. POWER AMPLIFIERS

The main amplifier section of Fig. 27-1 is redrawn in Fig. 27-9. The function of this circuit is to develop audio power. Let's go into the why and how of audio power generation. The power is needed to drive loudspeakers. Speakers, connected to the secondary of the output transformer, draw power from the power output tube, V_3 . It is in the power output stage V_3 that the audio power is generated. Sometimes more than one power tube is used in the output stage. All the tubes in the output stage assist in generating the power, but no other tubes in the amplifier do. The voltage amplifier, V_2 , controls the power that is generated in the output stage by means of an audio voltage. This audio voltage is produced at the plate of the voltage amplifier, V_2 ; then it

is coupled to the grid of the power tube, V_3 , where it controls the power tube and causes it to generate audio power.

This description may seem strange. So let's ask and answer a few questions to help clear things up. First, what is the basic difference between V_2 and V_3 that makes one stage a voltage amplifier, the other a power amplifier? The answer is that the power tube, V_3 , is capable of handling much more plate current than the voltage tube, V_2 . Both amplifiers work in the same basic way — a voltage variation (audio) at the grid produces a voltage variation (usually larger) at the plate. However, the voltage variations at the plate of V_3 are accompanied by variations in plate current that are much larger than the variations in plate current that accompany plate voltage variations in V_2 . Power is a combination of voltage and current ($P = EI$). Since the audio current in V_3 can be greater than that in V_2 , V_3 can deliver much more audio power than V_2 .

How does the audio voltage from V_2 control V_3 ? It does so by varying the voltage charge on the grid of V_3 and thereby varying the V_3 voltage-current output, which is audio power.

Doesn't V_2 have a voltage-current output and therefore a power output? Yes, but the current output is small and therefore the power output is small. It is the voltage output of V_2 that is needed to drive V_3 .

Why is it only audio voltage, and not audio voltage plus current (power) that is needed to drive V_3 ? No current can flow in the grid circuit of V_3 because V_3 is operated class A. Therefore, no current need be provided by V_2 for the grid circuit of V_3 .

Can an audio power tube be operated other than class A, and, if so, is audio power consumed in its grid circuit? Yes, grid current flows for part of an audio cycle whenever more audio voltage is applied to a grid than the tube can handle with the d-c bias that is present. Applying more signal than a tube can handle is called overdrive. Overdrive is commonly, though not nec-

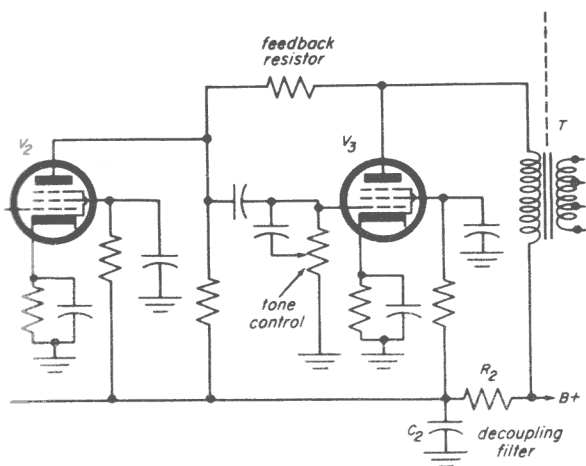


Fig. 27-9

essarily, used when an audio power tube is biased class B. Then power is consumed in the grid circuit because grid current flows during part of the audio cycle. The power consumed in the class B grid circuit must be supplied by the voltage amplifier. With class B operation, the voltage amplifier tube must be able to handle more plate current than with class A operation. Such voltage amplifiers deliver power but do not deliver nearly as much power as does the power-output stage. And the power they deliver is consumed in the grid circuit of the power output stage; it is not the power delivered to the speakers. It is still audio voltage applied to the output tube grid circuit (not the power consumed in the grid circuit) that controls plate power output from the power stage.

A power-amplifier tube works in the same basic way as a voltage-amplifier tube; that is, a grid-voltage variation produces a plate-voltage variation with both circuits, and the only difference between them is greater plate current variations in the power tube. Since this is so, and granting that a voltage tube cannot be used as power amplifier, — can't a power tube be used as a voltage amplifier? The answer is, yes. And this is an interesting question, because it brings to light another difference between voltage and power tubes, other than current handling ability. The difference is gain. Voltage tubes have the ability to amplify more than power tubes; voltage tubes have more gain. The difference in current handling ability and gain lie in the mechanical construction of the tubes, not in the circuits in which the tubes are employed.

Where only a small input signal voltage is available, it is best to use a voltage tube as voltage amplifier. On the other hand, power tubes are capable of producing larger signal output voltages at their plates, provided a large enough input signal is available to drive them. So, where large signal voltages are needed, power tubes are used as voltage amplifiers. For example, power tubes are sometimes used as voltage amplifiers to drive other power tubes used as power amplifiers, when the grid drive voltage requirements of the power amplifiers is very high.

This is a good place to list a few general statements about the grid-drive voltage requirements of power amplifiers:

1. In general, the more power output a power amplifier is capable of, the more grid-drive voltage it requires.

2. Triodes require more grid-drive voltage than pentodes operating under similar conditions.

3. A power amplifier with negative feedback requires more drive voltage than the same tube without feedback. (Negative feedback can make a pentode behave like a triode in this respect and in other respects, too).

4. More drive voltage is required in class AB than for the same tube in class A, and more drive voltage is required in Class B than either Class AB or A.

If a power amplifier is an audio power generator controlled by an audio voltage, can it have gain? A power amplifier can have gain. Gain for a power amplifier means the amount of signal voltage you get out for a certain grid drive voltage you put in. Gain for a power amplifier is expressed in terms of voltage, just as for a voltage amplifier.

However, power-delivering ability is often more important than gain. For example, a power stage provided with negative voltage feedback has less gain than the same stage without feedback; more input voltage must be applied to the grid with feedback to get the same output voltage at the plate as without feedback. Yet, when the input (drive) voltage requirement with feedback is met, identical power output is obtained under more desirable conditions.

Does more gain always result in more power output as well as more voltage output for the same voltage input? It does if the same tube is involved in each case. (For example, the same tube with and without feedback). Also, one way of explaining why a pentode circuit can have more power output than a triode with the same drive-voltage

input and the same plate-current handling ability is to point out that pentodes have more voltage gain. More gain and more power output for a given voltage input usually go together. Nevertheless, the two qualities are not, strictly speaking, the same thing. The measure of power output versus voltage input is called *power sensitivity*.

A voltage amplifier can be described in terms of the following:

1. Voltage gain
2. Maximum voltage output

A power amplifier stage can be described in terms of the following:

1. Voltage gain
2. Power sensitivity
3. Maximum power output

This brings us to the final question. If gain for a power amplifier is in terms of voltage, what is the meaning of gain for power amplifiers given in the decibel system that is basically a ratio of two powers? The answer is that the so-called decibel gain of a power amplifier is a way of rating the power sensitivity of the amplifier. Let us see how decibels can be used for this purpose. Moreover, let us take up the subject of power amplifier rating in general and learn what various ratings mean and how they are related to each other.

Amplifier Ratings. A list of amplifier ratings that are related to each other and must be studied together is as follows:

1. Voltage gain
2. Power sensitivity
3. Decibel power gain
4. Input impedance

Voltage gain, you remember, is how much voltage you get out for a certain amount of voltage you put in. It is the following ratio:

$$\text{Voltage Gain} = \frac{E_{\text{out}}}{E_{\text{in}}}$$

The word *gain* is used when both quantities in the ratio are the same kind. In this case, both are voltage (E).

Power sensitivity is how much power you get out for a certain amount of voltage you put in. It is the ratio:

$$\text{Power Sensitivity} = \frac{P_{\text{out}}}{(E_{\text{in}})^2}$$

The word *sensitivity* is used when the two quantities in the ratio are different. In this case, one is power and the other is voltage.

Although the power sensitivity is defined as the ratio of the power to the input voltage squared, as given above, it is usually not stated this way in literature giving the specifications of amplifiers. The sensitivity is usually given as the voltage required to produce the maximum output. For example, an amplifier might be said to have a sensitivity of 1 volt input for 10 watts output. This tells that the maximum output is 10 watts, and that 1 volt input signal is needed to produce this maximum output power. It should be remembered that the power output is related to the square of the input voltage; thus reducing the input voltage in this case to 0.5 volt will produce 0.25 of the power or 2.5 watts.

Power gain is the ratio:

$$\text{Power Gain} = \frac{P_{\text{out}}}{P_{\text{in}}}$$

And decibel power gain is the decibel equivalent to this ratio as follows:

$$\text{db power gain} = 10 \log \frac{P_{\text{out}}}{P_{\text{in}}}$$

If the rating of the amplifier is not given in terms of sensitivity, but in terms of power gain, then it is necessary to know the input impedance of the amplifier in order to compute the voltage input needed. For example, suppose we know that an amplifier has a power gain of +40 db, an input impedance of 1,000 ohms and a maximum power output of 10 watts. For one milliwatt input

$$E = \sqrt{PR} = \sqrt{1 \times 10^{-3} \times 1,000} = 1 \text{ volt}$$

so that one volt input is required.

Let's consider the list of ratings to be studied together and draw some conclusions. First, we can conclude that since P_{in} and R_{in} together determine E_{in} , power sensitivity and power gain with input impedance are two ways of giving the same rating.

To illustrate, we can state the power sensitivity of the amplifier directly by saying that *the amplifier produces 10 watts output at one volt input*. If we wish to speak of power gain instead of power sensitivity, we state the following:

Input impedance is 1,000 ohms

Power output is 10 watts

Power gain is 40 db

Since the direct sensitivity rating seems simpler, you may wonder why the power-gain - input-impedance rating is used at all. Perhaps we can explain with the aid of an account of what may very well have taken place in the early days of radio.

Many years ago, in the early days of radio, a radio pioneer may have had an old fashioned microphone and connected it to a pair of earphones. When the radioman talked into his mike, he could not hear himself in his phones. He thought about the problem and decided that the mike's power output was not enough to operate the phones. His mike was of a kind that generates an electric power signal corresponding to the sound input. He decided to get more power out of the mike. In order to do so, he arranged to terminate the mike with 1,000 ohms, an impedance that matched the mike impedance. He knew that you get maximum transfer of power into a matched load. This done, he was able to measure an increase in the power delivered by his mike. But it still would not operate the phones. The mike output was about 0.000001 watt. So he decided to amplify the power.

He knew that he would have to find out how much voltage he had available from his mike to operate a power amplifier. This was simple to find. One-millionth of a watt,

dissipated in 1,000 ohms, produces 0.033 volts. Since the phones required about 5 mw, he decided to use a 5 mw amplifier with 0.03 volts sensitivity in order to amplify the millionth of a watt to 5 mw.

Notice that we speak of the amplifier as having a sensitivity of 0.03 volts. This is the first time we have spoken of sensitivity in this way. Up to this point we have spoken of sensitivity as the *ratio* of a particular known voltage input to a particular known power output. However, sometimes it is convenient to specify sensitivity in terms of voltage alone, just as we have done in the paragraph above. Sensitivity expressed in this way means that for the voltage input specified, the amplifier will deliver maximum power output, whatever that power output may be. For example, even if we had not stated that the power output of the amplifier we are discussing is 5 mw, we could have specified its sensitivity as being 0.03 volts. This means that an input of 0.03 volts will result in a maximum power output.

Let's get back to the problem of our pioneer radioman. He thought of the problem in terms of power. He had a mike that delivered power and a set of phones that consumed power. His problem was to increase the power delivered by the mike to make it operate the phones. Therefore, he rated his amplifier in terms of power gain. He said to himself, "Now I've got a mike with an output of -30 dbm (0.000001 watt is -30 db compared to the standard of 1 mw) and an amplifier with a power gain of +37 db.

This is because:

$$\begin{aligned} \text{db} &= 10 \log \frac{P_2}{P_1} \\ &= 10 \log \frac{5 \text{ mw}}{0.000001 \text{ w}} \\ &= 10 \log 5,000 \\ &= 37 \end{aligned}$$

That will give me +7 dbm output power, or 5 mw, which is what I need for the phones."

This brings us to our second conclusion, which is that a particular decibel power gain holds true for only one value of input impedance. In the case of the amplifier we just discussed, the gain was +37db with 1,000 ohms input impedance. Let us figure out what the gain would be for an input impedance of 10,000 ohms. We know the sensitivity is 0.033 volt input for 5 mw output. A 0.033-volt input across 10,000 ohms makes an input power requirement of 0.1 microwatts or -40 dbm. A -40 dbm input and a +7 dbm output make a gain of +47 db. Note that changing the input impedance from 1,000 ohms to 10,000 ohms increases the power gain rating from 30 db to 40 db. Of course, the amplifier is exactly the same in each case and has the same sensitivity. The difference is that less power is required to develop 0.03 volts across 10,000 ohms than across 1,000 ohms, and therefore less input power is required into 10,000 ohms for the same amplifier output power. Therefore, the amplifier has greater power gain at 10,000 ohms input.

But our old-time radioman did not think of how low input resistance affects power gain. He had to match his mike with 1,000 ohms, and that was that. It was convenient to be able to add the dbm power output of his mike to the db gain of his amplifier and find the power output that would be produced.

Currently microphones are not terminated in their rated impedance. Instead, they are terminated in an impedance much larger than their rated impedance. As a result, the actual voltage output is about twice as large as it would be if the mikes were terminated in their rated impedance. This means that the actual voltage output is about twice as great as the voltage value you come up with when you use dbm rated power output and rated output impedance to figure out the corresponding voltage. Nevertheless, amplifier power gain continues to be rated as if the microphones were terminated in their rated impedance and the output voltage were half what it really is. For this reason, when a mike has an output rated in dbm, this can still be

added to the db power gain rating of an amplifier to find the power output.

However, when a microphone output is rated in dbv this cannot be done, not because of termination considerations but because dbv and db gain are different units. The former is a voltage unit; the latter is a power unit. To add them together would be like adding apples to oranges. Such an addition cannot be carried out. Unfortunately, most low and medium power PA amplifiers are rated in terms of power gain (high-power amplifiers are rated directly in sensitivity) while the high-impedance microphones usually used with these amplifiers are rated in dbv output. You must learn how to figure out what power output the amplifiers will deliver when supplied with an input equal to the rated dbv output of these mikes. There are several ways to go about the task.

One way is to convert the power-gain - input-impedance rating of the amplifier to a sensitivity rating and convert the rated dbv mike output to volts. A second way is to convert the dbv rating of the mike to a dbm rating based on the amplifier's rated input impedance. A third way is to convert the power-gain - input-impedance rating of the amplifier to a sensitivity rating in dbv. Then the dbv mike output can be added to the dbv sensitivity of the amplifier to find what power output will be produced.

Let us take a typical amplifier as an example and use all three methods. But first, let us get one thing straight. The sensitivity rating that will result from converting power-gain - input-impedance rating is nominal and not the real sensitivity rating. That is, it is not the sensitivity rating that the manufacturer would have put upon the amplifier had he chosen to do so instead of rating the amplifier in terms of power gain. He would have *measured* the voltage input required to produce maximum power output. As was pointed out earlier, db power gain is computed as if the input voltage were half what it actually measures, because it is twice what it actually would measure if microphones were terminated in a matched load. Also, the procedures in the examples to

follow give exact results when applied to high-impedance magnetic microphones rated in dbv output across a rated impedance like 40,000 ohms. However, the results are only approximate when applied to high-impedance crystal and condenser microphones or any other microphones rated in dbv open circuit output without any rated output impedance. When the dbv output is given for a rated impedance, it is about half or 6 dbv less than the actual output. This corresponds to the conditions under which the power gain of an amplifier is computed. On the other hand, crystal mikes are rated in terms of actual unterminated output voltage, and this does not correspond with conditions under which power gain is computed. In general, when the high-impedance input of the amplifier is low (say 0.5 megohm), the error from applying the following procedure to crystal mikes is greatest. Amplifier input impedances range to as high as 10 megohms, but 0.5 megohms is common in PA equipment at the present time.

Assume that an amplifier has the following rating:

Power output: 50 watts

Power gain: 125 db.

Input impedance: 1/2 megohm

To this amplifier, we want to connect a high-impedance magnetic microphone rated at -50 dbv output across 40,000 ohms. Now let us apply our first method. First we will convert both mike output and amplifier sensitivity to volts. To convert the amplifier rating to volts sensitivity proceed as follows:

1. Convert 50 watts to dbm with the decibel tables in Theory Lesson 26:

$$50 \text{ watts} = +47 \text{ dbm}$$

2. Find the dbm input by subtracting algebraically the db gain from the dbm output:

$$\text{input} = 47 - 125 = -78 \text{ dbm}$$

3. Convert the dbm input to watts using the tables:

$$-78 \text{ dbm} = 16 \text{ microwatts}$$

4. Find the voltage across 1/2 megohm when 16 microwatts is dissipated:

$$P = \frac{E^2}{R}$$

$$E^2 = PR$$

$$= 16 \times 10^{-12} \times 0.5 \times 10^6$$

$$= 8 \times 10^{-6}$$

$$E = \sqrt{8 \times 10^{-6}}$$

$$E = 0.0028 \text{ volts}$$

The nominal sensitivity of the amplifier is stated as: *0.0028 volts input produces 50 watts output*. Now, to convert the dbv output of the mike to volts, look up -50 dbv in the tables. You see that -50 dbv = 0.0032 volts. So the mike is rated to deliver $0.0032 \div 0.0028$ or 1.14 of the required input voltage and will produce $(1.14)^2$ of the 50 watts output, or 65 watts. Since the maximum power output is only 50 watts, we know that the mike is capable of driving the amplifier to more than full rated output.

Now let us do the same thing using the second method. You can convert the -50 dbv output rating of the mike to dbm into 0.5 megohms by subtracting the following correction factor from the dbv rating:

$$10 \log \frac{R_2}{R_1}$$

R_2 in this case is 0.5 megohm. For converting dbv to dbm, R_1 is always 1,000 ohms.

$$\frac{R_2}{R_1} = \frac{500,000}{1,000} = 500$$

To find $10 \log 500$, look up 500 in the *Ratio* column Table B of Lesson 26 and get the answer from the (*db Power Ratio*) column. The power db equivalent of 500, which is $10 \log 500$, is shown by the table to be +27 db. Subtracting +27 db from -50 dbv we get -77 dbm.

The -77 dbm output rating of the microphone can be added directly to the power gain of the amplifier to find the dbm amplifier output. To do this, we add -77 dbm to +125 db. The result is an amplifier output of +48 dbm. Looking in the tables we find that +48 dbm equals 63 watts. Last time we got an answer of 65 watts. The difference is due to approximations made when consulting the table.

The third method is to convert the amplifier rating to a sensitivity rating in dbv. You do this as follows:

1. If an amplifier has a gain of +125 db and an output of 50 watts, its required input is -125 db below 50 watts. We know that 50 watts is +47 dbm, or +47 db above 1 mw. Therefore, -125 db below 50 watts must be -125 + 47, or -78 db below 1 mw, or -78 dbm.

2. We can convert -78 dbm in 0.5 megohms to dbv across 0.5 megohm by adding the following correction factor:

$$10 \log \frac{R_2}{1,000}$$

where

$$R_2 = 0.5 \text{ megohm}$$

We have found that this correction factor is +27 db. So -78 dbm + 27 db = -51 dbv. The dbv sensitivity of the amplifier is -51 dbv input for +47 dbm output. A change in the input, expressed in dbv, produces the same numerical change in the dbm output. Since our mike has -50 dbv output, it has 1 dbv more than is necessary, or enough to cause an output of 48 dbm (approximately). Of course, the amplifier can deliver only 47 dbm, but if it could deliver 48 dbm this would equal a power output of 63 watts, which is the answer we got last time.

There is one important thing to learn about input impedance ratings. Rated impedance is not always actual input impedance. Rated input impedance is the quantity to be used in the calculations we have been describing. And amplifier rated impedance should correspond to the rated impedance of the mike that you connect to the amplifier's input terminals. But the actual input impedance of the amplifier is often higher than the rated impedance. To understand, let us analyze an amplifier designed to have two alternate input impedances. Suppose the amplifier were rated as follows:

Power output:

50 watts

Input impedance:

#1 high is 0.5 megohm

#2 low is 200 ohms

Gain:

High impedance is 125 db

Low impedance is 114 db

The input circuit is shown in Fig. 27-10. The rated input impedance is normally 0.5 megohm. To obtain a rated input impedance of 200 ohms, it is necessary to plug in the transformer (T). The 0.5-megohm rated input impedance is an actual impedance. It is determined by the resistor, R . The 200-ohm rated impedance is actually greater than 200 ohms. Let us figure out what it actually is. First, we figure out the high-impedance sensitivity. Since the high-impedance channel of this amplifier is identical to the amplifier considered previously, we know the high impedance sensitivity is 0.0028 v. Second, then we figure out the 200-ohm sensitivity as follows:

1. 50 watts output = 47 dbm

2. gain = 114 db

3. Therefore input

$$\text{power} = 114 \text{ db} + 47 \text{ dbm} = -67 \text{ dbm}$$

$$= 2 \times 10^{-10} \text{ watts}$$

4. And from $P = \frac{E^2}{R}$, the input

voltage is as follows:

$$2 \times 10^{-10} = \frac{E^2}{200}$$

$$E^2 = 400 \times 10^{-10} = 4 \times 10^{-8}$$

$$E = \sqrt{4 \times 10^{-8}} = 2 \times 10^{-4}$$

$$= 0.0002 \text{ volts}$$

The sensitivity of the amplifier is 200 ohms with an input voltage of 0.0002 volts. This means that if the 200-ohm microphone that is to be connected at the low impedance input were terminated, it would have an output of 0.0002 volts across its 200 ohms. And with this condition, the transformer (T) turns ratio is such that the secondary voltage will be 0.00028 volts—the nominal high-impedance sensitivity voltage. With this information, we can figure out the turns ratio. The formula for transformer turns ratio, you remember, is as follows:

$$\frac{E_2}{E_1} = \frac{N_2}{N_1}$$

Making use of this formula, we have:

$$\frac{N_2}{N_1} = \frac{0.0028}{0.0002} = \frac{14}{1}$$

We know that the turns ratio is 14 to 1. We also know that the secondary impedance is 0.5 megohms. With this information, we can use the transformer impedance ratio formula to find the actual primary or input impedance (Z_1) as follows:

$$Z_1 = \left(\frac{N_1}{N_2} \right)^2 \times Z_2 = \left(\frac{1}{14} \right)^2 \times 500,000$$

$$= \frac{1}{196} \times 500,000 = 196 \sqrt{500,000}$$

$$= 2,551 \text{ ohms}$$

So you see that while the low impedance input is rated at 200 ohms, it is actually 2,500 ohms. This means that a 200-ohm low impedance microphone connected to the low impedance input would be unterminated (in its rated impedance) and unmatched. In turn, this means that such a mike would deliver about 6 db of voltage more than its dbm output rating indicates, or about two times the output voltage indicated by its dbm output rating.

The 200-ohm mike is terminated in 2,500 ohms, an impedance $12\frac{1}{2}$ times its rated impedance. This is equivalent to terminating a 40,000-ohm mike in 0.5 megohm, which is $12\frac{1}{2}$ times 40,000 ohms. In fact, if we use the turns ratio formula to find the secondary impedance of T with the 200-ohm mike connected to the primary, we come up with a secondary impedance of 40,000 ohms, and this is loaded by the 0.5-megohm resistor.

Power Circuits. At the start of this booklet, we pointed out that some PA amplifiers are just like radio audio circuits but that others are capable of delivering much more power. Let us briefly consider the circuits employed to provide higher power output. A designer can obtain high power output by doing any one of the following:

1. Using a high-power type tube in the output stage.
2. Connecting two output tubes in push-pull.

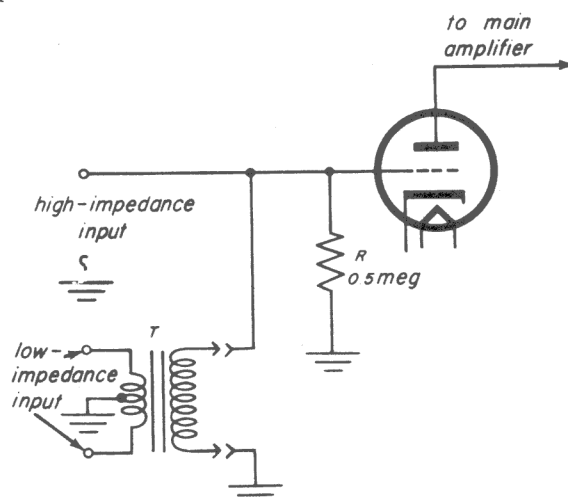


Fig. 27-10

3. Connecting four tubes in push-pull parallel (two tubes in parallel and in push-pull with two other tubes in parallel).
4. Operating the output tubes in class AB or B instead of class A.
5. Adopting a combination of two or more or even all four of the above measures.

Figure 27-11 shows some representative power amplifier circuits in the order of their power-output capabilities. A single-ended class A triode provides about the least power output of all power amplifier circuits. Normally, a single voltage amplifier can drive the triode to full power output with phonograph input. With microphone input, a voltage preamplifier is needed. A single class A pentode provides somewhat more power output, but no more voltage amplifier tubes are needed. When two power tubes are connected in push-pull, a phase inverter is required in addition to the voltage amplifiers. And since two tubes in push-pull require twice as much drive voltage as the same tubes in single-ended operation, another voltage amplifier tube may be required. Whether or not the additional voltage amplifier is required depends upon whether or not the necessary voltage gain and magnitude of voltage output can be obtained from a single voltage-amplifier tube. Connecting tubes in parallel does not increase the grid-drive voltage requirements, so four tubes in push-pull parallel require no more voltage amplification ahead of them than do two tubes of the same type in push-pull. Changing the class of operation of power tubes from class A to class AB or class B increases the grid-drive requirements and thus necessitates additional voltage amplifiers. Changing from class A to class AB or B, and the use of pentodes in general, increases distortion; this necessitates negative feedback and results in increased drive requirements.

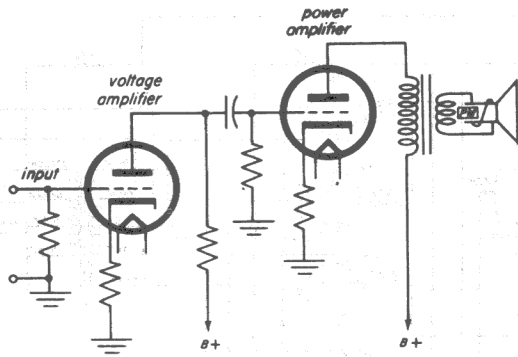
Single-ended power amplifiers are usually operated class A. Push-pull power amplifiers can be operated class AB without a significant increase in distortion. In former times, class B operation and class AB operation with overdrive were reserved for installations where small equipment size and low power-supply current drain are more

important factors than freedom from distortion. A mobile PA installation is an example of an installation where size and power supply drain are more important than freedom from distortion. Recently, circuits have been developed that make it possible to operate power tubes in bias classes other than class A without distortion.

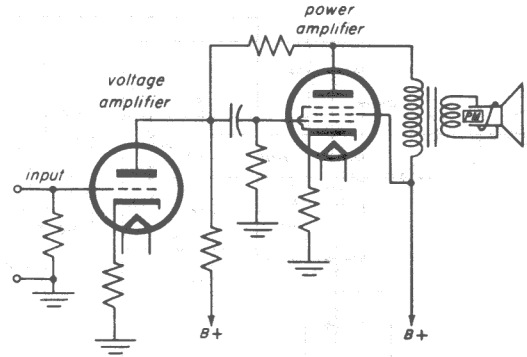
Power-supply current drain is an important consideration affecting the design of power amplifiers. In general, the types of tubes capable of most power output draw the most current. Of the various power tube types that are listed in the *RCA Receiving Tube Manual*, those that deliver more power differ from those that deliver less power principally in the amount of power-supply current drawn. These tubes all have similar power voltage requirements (B+), ranging to approximately 300 volts. However, where more power is required than can be supplied by the receiving type of tube, special tubes not listed in the receiving tube manual are employed.

These tubes may require a power voltage that is much above 300 volts. An example of a tube that is not a receiving tube type and which is a widely used power amplifier in PA equipment is type 807. This tube requires about 600 volts B+. It is classified as a transmitter-type tube. Electrically, it is very similar to the 6BG6 beam power tube used as a horizontal-scanning power amplifier in TV receivers. The 6BG6 is listed in receiving tube manuals.

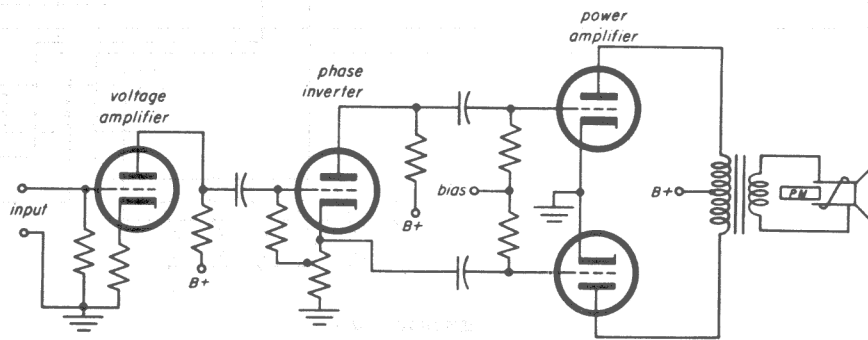
Recent Trends. Recently developed high fidelity audio-power amplifier circuits depart from the fundamental circuits shown in Fig. 27-11 and cannot be adequately described, as the fundamental circuits are described, by such terms as push-pull, class AB, push-pull-parallel, and etc. As a result they are given names—sometimes in honor of the developer of the circuit, sometimes with reference to the way in which the circuit operates. The Williamson amplifier and the McIntosh amplifier are examples of the newly developed high-fidelity audio-power amplifiers. A study of these circuits comes under the heading of high-fidelity audio rather than PA systems and is beyond the scope



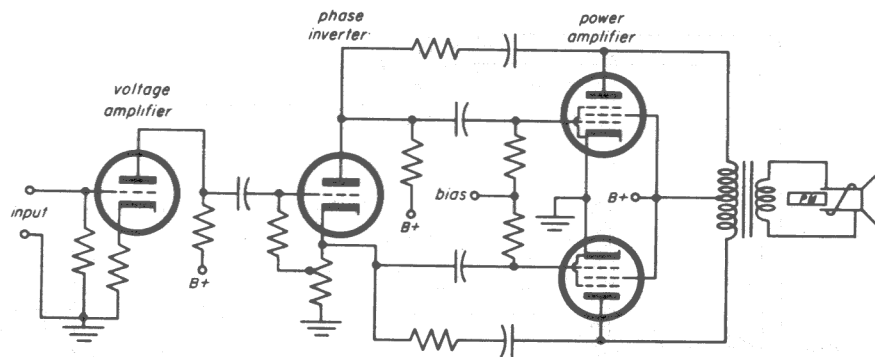
single-ended triode power amplifier



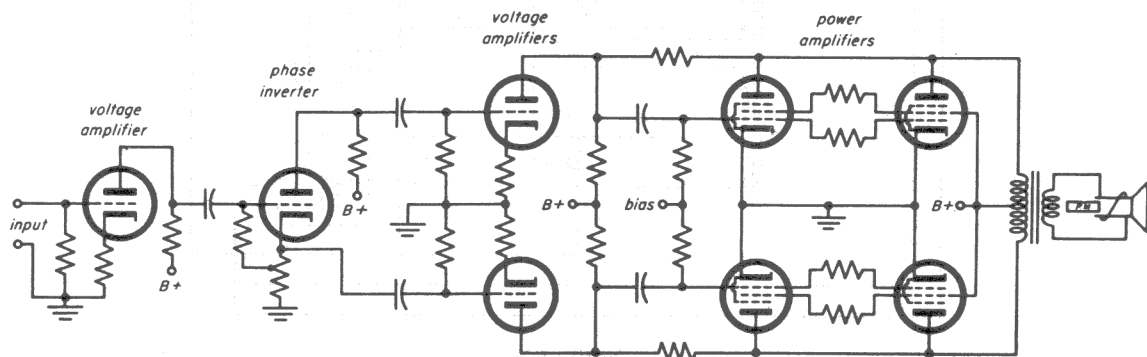
single-ended pentode power amplifier



push-pull triode power amplifier



push-pull pentode power amplifier



push-pull parallel pentode power amplifier

Fig. 27-11

TABLE B

Power output:	100 watts rated, 200 watts on peaks
Frequency response:	± 0.5 db 20 to 50,000 cps
Harmonic distortion:	less than 0.5% at 100 watts output
Intermodulation distortion:	less than 1.2% at 100 watts output
Hum and noise level:	85 db below 100 watts output
Output impedances:	4 ohms, 8 ohms, 16 ohms, 600 ohms, and a 70-volt constant-voltage output
Input impedance:	250,000 ohms
Sensitivity:	1.25 volts rms for 100 watts output

of this booklet. However, high-fidelity, high-powered audio amplifiers are coming into more and more general use in PA equipment. So let us briefly examine the circuit of a high power, high-fidelity amplifier intended for PA applications. The amplifier is manufactured by the Electro-Voice Co. of Buchanan, Michigan. It has the specifications shown in Table B.

A schematic of the circuit is shown in Fig. 27-12. It consists of a triode voltage amplifier (V_{3A}) driving a phase inverter (V_{3B}) which drives two more amplifier tubes (V_{4A} , V_{4B}), each of which drive two power tubes connected in parallel.

Four power tubes are employed in a special kind of push-pull parallel circuit called a Wiggins Circlotron® circuit. A simplified version of the Wiggins power output circuit is shown in Fig. 27-13. It is a balanced push-pull circuit that works something like the circuit of the VoltOhmyst® amplifier described in Service Practices 25. With no input signal, the circuit is balanced. No current flows in the primary winding of the output transformer while the circuit is in the balanced condition. As a cycle of incoming audio signal goes positive (or negative) from zero, the instantaneous voltage on the grids of the other parallel pair of tubes decreases. Thus, the circuit is unbalanced by the incoming signal and a current flows in the primary winding of the out-

put transformer. The current in the transformer follows the variations of the input signal and delivers a corresponding output power signal to the loudspeaker.

Because no d-c current flows in output transformer, and also because of other more complex considerations involved in the design of the circuit, an unusually efficient transformer can be manufactured to fulfill the circuit's requirements. This transformer contributes to the excellent frequency response and low distortion of the amplifier. (In general, an output transformer is one

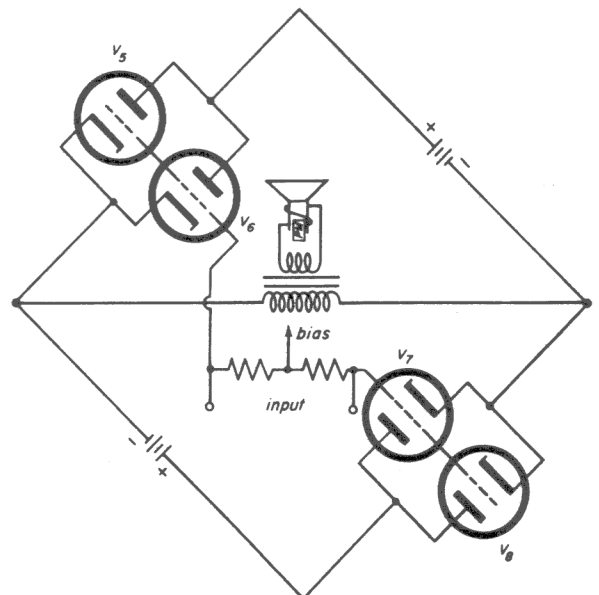


Fig. 27-13

of the weakest links in the chain of circuit components making up a high-fidelity amplifier.)

Although the plates and cathodes of each parallel pair of tubes are separated by d-c power supplies (represented by batteries in Fig. 27-13), the tubes are effectively connected together from the standpoint of signal voltage at the audio frequencies. So each half of the output transformer primary winding is connected to the plate of one pair of tubes and to the cathode of the other pair of tubes simultaneously. This arrangement makes the transformer primary a plate load for one pair of tubes while it is a cathode load for the other tubes. A large cathode load such as this causes much degeneration, which contributes to make possible the high-fidelity features of the amplifier. The degeneration is so great that the gain of the output circuit is 1 to 1, or no gain at all. Thus, the output stage has very low sensitivity. Three voltage amplifier tubes are needed to make the sensitivity of the entire amplifier what it is. The manufacturer's specifications give the sensitivity at 1.25 volts for full rated output. This is not adequate for phonograph input. Phonograph input requires a sensitivity of about 0.5 volts. The amplifier is designed for use with a separate mixer amplifier having 1.25 volts high-impedance output.

27-4. OUTPUT CIRCUITS

You recall that we described the output section of Fig. 27-1 earlier. The circuit is redrawn in Fig. 27-14. This is probably the simplest form of output circuit found in PA equipment. The circuit has provision for connecting a 4-, 8-, or 16-ohm speaker load. Figure 27-14 depicts various combinations of 8-ohm speakers properly connected to the output circuit. In PA applications, 8-ohm speakers are widely used. Most PA equipment has provision for connecting 4-, 8-, and 16-ohm speaker loads. However, when long cables are to be used to connect distant speakers, an impedance of 16 ohms is too low. To understand, examine the circuit of Fig. 27-15. If the connecting wire is long enough, it will have a large impedance compared to the 8-ohm speaker impedance.

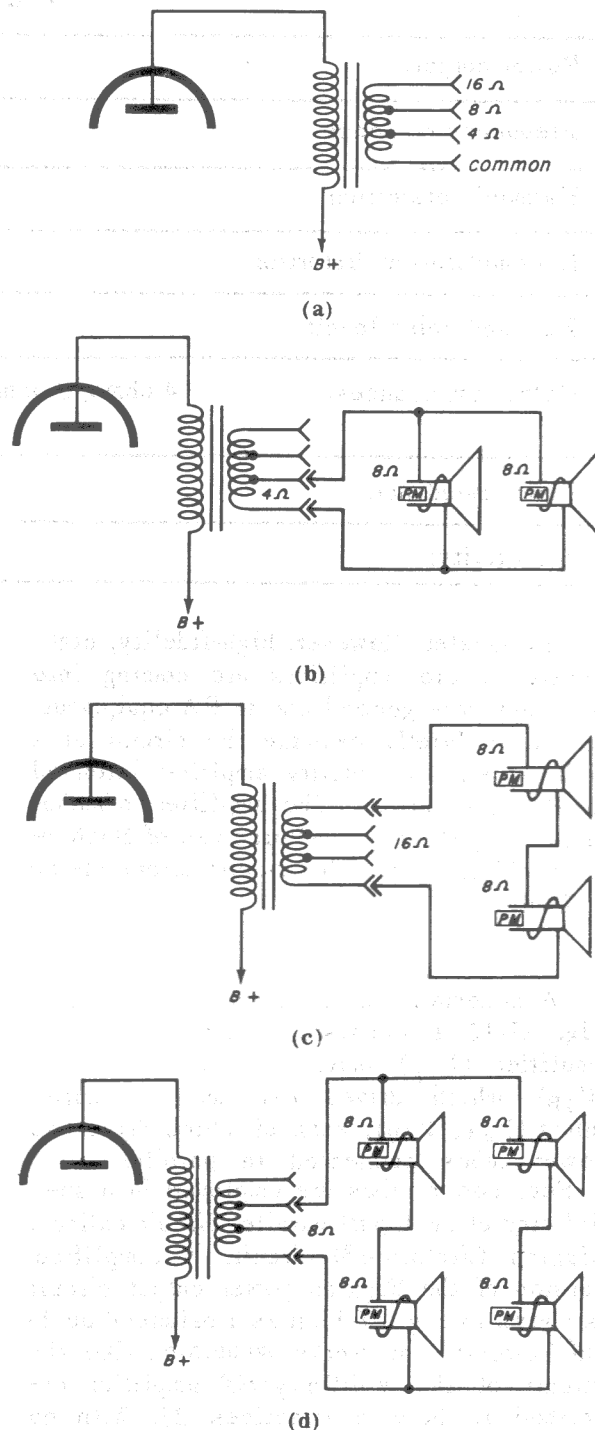


Fig. 27-14

Much of the signal power will be lost in the connecting wire.

The way in which this limitation may be overcome is shown schematically in Fig. 27-16a. Here, the output circuit has provision for connecting a 500-ohm load. A 500-ohm speaker load at the end of a long cable is connected to the 500-ohm amplifier

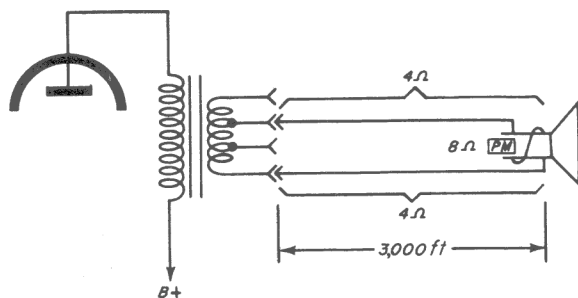


Fig. 27-15

output tap. A very long wire can be used. The wire will not have an impedance that is large compared to 500 ohms. Of course, the heavier the connecting wire, the less the power loss. Here are a few examples to give you an idea of what is involved.

If an 8-ohm speaker load is connected to an 8-ohm amplifier output tap by means of 250 feet of connecting cable made up of B&S No. 14 copper wire, the signal power loss is 15%. No. 14 wire is rather heavy. It is the wire size used often in BX cable for 110-volts electric-lighting circuits. If No. 18 wire is used to connect an 8-ohm speaker to an 8-ohm output tap, only 100 feet causes a 15% power loss. This No. 18 stranded wire is used in household electric extension cords, for lamp cords. No. 18 solid wire is used in the B+ wiring of home radio receivers. If No. 14 wire is used to connect a 500-ohm speaker load to a 500-ohm amplifier output tap, 5,000 feet of wire causes a power loss of only 5%. If No. 18 wire is used at the 500-ohm impedance level, 2,000 feet causes a 5% loss.

So you see that the 500-ohm impedance level is best for long speaker-connecting cables. Most PA amplifiers have provision for connecting a 500-ohm speaker load (or some other high impedance value) as well as the 4-, 8-, and 16-ohm speaker loads. Other values of high-impedance speaker load in common use are 600 ohms and 250 ohms.

Although the high-impedance level is very desirable for long lines, speakers cannot readily be built with a 500-ohm voice coil impedance. Instead, transformers are used at the speaker location to raise the speaker voice coil impedance (4, 8, or 16 ohms) to the 500-ohm level. This is illustrated in Fig. 27-16b.

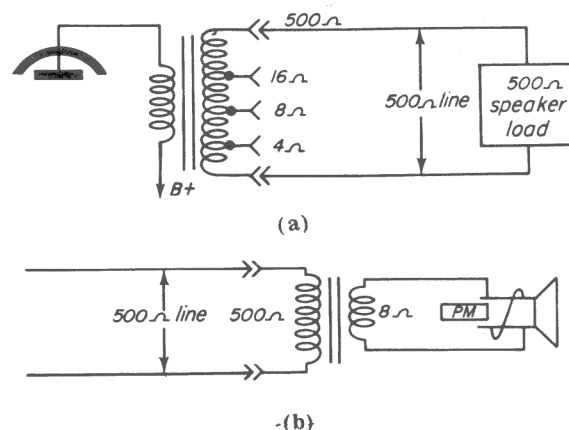


Fig. 27-16

Constant-Impedance System. Let's discuss the way to connect several speakers to the 500-ohm line. The system we are going to describe is called a constant-impedance system. This means that the total impedance of all the speaker loads together, no matter where they are located on the 500-ohm line, must equal 500 ohms at all times. If a speaker is added, or if one is removed, the remaining speakers must be rearranged to have a total impedance of 500 ohms. If one speaker is to get more signal power than another, this must be provided for without changing the total impedance of 500 ohms.

To begin, suppose we want to connect four 8-ohm speakers to the end of a 500-ohm line. We can connect the speakers either in series or in parallel. The parallel arrangement is best. In general, it is good practice never to connect more than two speakers in series. Connecting too many speaker voice coils in series results in a large total inductance (inductance in series adds). Strong, sharp peaks in the sound signal represent high momentary voltage peaks. The larger the inductance, the larger the value of momentary voltage. If the voltage gets large enough, it causes arc-over. The arcing can damage the voice coil. Keeping the inductance low by using parallel rather than series voice coil combinations is a precaution against arc-over. Series-parallel combinations are satisfactory if no more than two speakers are in any one series string.

When we connect four 8-ohm speakers in series-parallel (Fig. 27-17a), the total impedance is 8 ohms. We need a transformer to match 8 ohms to 500 ohms. Transformers of

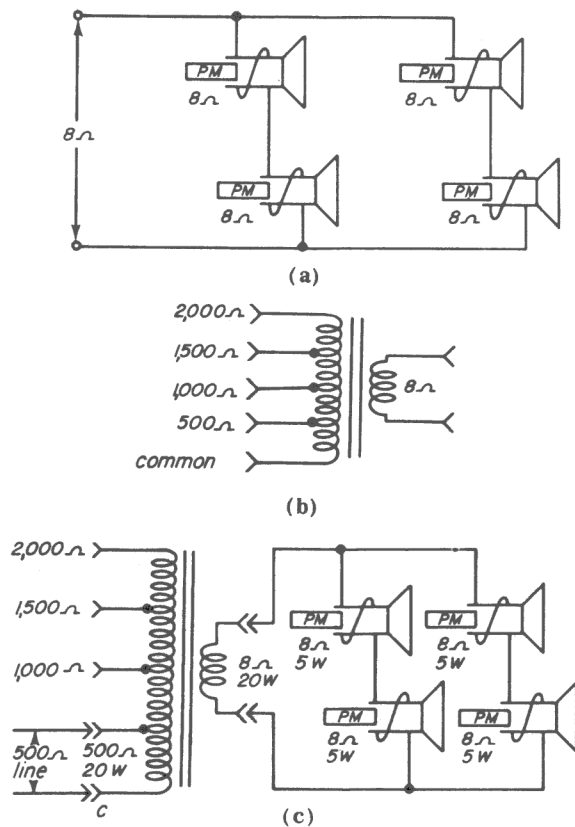


Fig. 27-17

this kind are called *speaker-to-line matching transformers*. They are rated in terms of power-handling ability, voice-coil impedance, and the several line impedances that can be matched to the rated voice-coil impedance. A suitable speaker-to-line transformer is shown schematically in Fig. 27-17b. It is rated at 20 watts and to match an 8-ohm voice coil impedance to either 500 ohms, 1,000 ohms, 1,500 ohms, or 2,000 ohms. We are interested in the 500-ohm tap. This can be connected to our 500-ohm line (Fig. 27-17c). Since our four speakers are connected in a way to make up a total impedance of 8 ohms, they can be connected to the 8-ohm terminals of the transformer and everything will match up.

The transformer is rated at 20 watts. Therefore, up to 20 watts can be delivered to the combination of four speakers. The power will divide equally among the four speakers because all the speakers have the same impedance (8 ohms). Each speaker can receive up to 5 watts (1/4 of 20 watts).

To take another example, suppose that we want to connect three speakers to the end of a 500-ohm line and that we want one of the three to receive twice as much signal

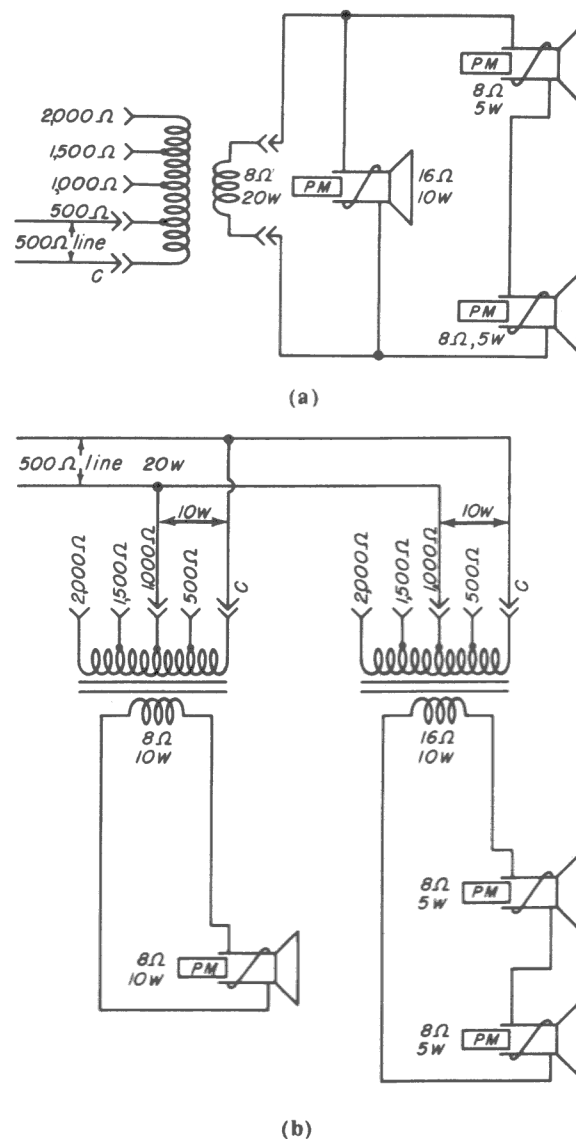


Fig. 27-18

power as either of the other two—the other two to receive equal power. We can accomplish our purpose by connecting two 8-ohm speakers in series in parallel with a 16-ohm speaker (Fig. 27-18a). This combination has a total impedance of 8 ohms. The power will divide in half between the 16-ohm speaker and the two 8-ohm speakers. The half going to the two 8-ohm speakers, in turn, divides in half between the two 8-ohm speakers. So 1/2 the total power goes to the 16-ohm speaker and 1/4 the total power to each 8-ohm speaker. In other words, the 16-ohm speaker gets twice as much power as either of the 8-ohm speakers. This is what we want.

Of course, the 16-ohm speaker must be rated to handle twice as much power as either 8-ohm speaker. We can use the same

transformer to match the 8-ohm total impedance to the 500-ohm line that we used in the previous example. This transformer is rated at 20 watts. So the 16-ohm speaker should be rated at 10 watts and the two 8-ohm speakers should each be rated at 5 watts.

Now suppose we want to connect three speakers to the end of a 500-ohm line, with the power distributed as before, but all we have is 8-ohm speakers—no 16-ohm speaker. If we connect one 8-ohm speaker in parallel with two 8-ohm speakers in series, we will not get the desired power distribution. Also, the combination makes up a total impedance of 5.3 ohms, and there are no standard voice coil-to-line matching transformers to match a 5.3-ohm voice coil impedance to anything. The transformers come rated to match the standard 4-, 8-, and 16-ohm voice coil impedances. Mostly, they come rated to match 8 ohms. What we can do is use two standard transformers as follows (Fig. 27-18b). We use one transformer rated to match a 16-ohm voice coil impedance to various values of line impedance. To the voice-coil winding of this transformer, we connect two 8-ohm speakers in series. We use another transformer rated to match an 8-ohm voice coil impedance to various values of line impedance and to this transformer we connect a single 8-ohm speaker. And here is where you find out about use of the impedance taps other than 500 ohms. We want to connect the line side of the two

transformers together in parallel. But we cannot connect the two 500-ohm taps in parallel because this makes a total of 250 ohms, whereas we want to have a total of 500 ohms to match the 500-ohm line. So we connect the two 1,000-ohm taps in parallel and thus obtain a total impedance of 500 ohms. The power divides equally between the two transformers so that half of the total available power is delivered to the speaker loads of each transformer. Half of the total power goes to the speaker connected singly to its matching transformer; but half of half, or one quarter of the total power, goes to each of the two speakers, which are connected in series across their matching transformer.

The higher-impedance taps are also useful when it is necessary to space speakers at widely separated intervals along a line. Suppose we want to connect speakers at 1,000-foot intervals along a 500-ohm line. We use a transformer for each speaker, and connect the transformers in parallel across the line as shown in Fig. 27-19. The total impedance across the line must be 500 ohms. So we use the 2,000-ohm tap of each of the four transformers. Four 2,000-ohm impedances in parallel total 500 ohms.

Constant-Voltage System. The constant-impedance system of matching loudspeakers to PA amplifiers has been in use for many years. But recently, a new system, called

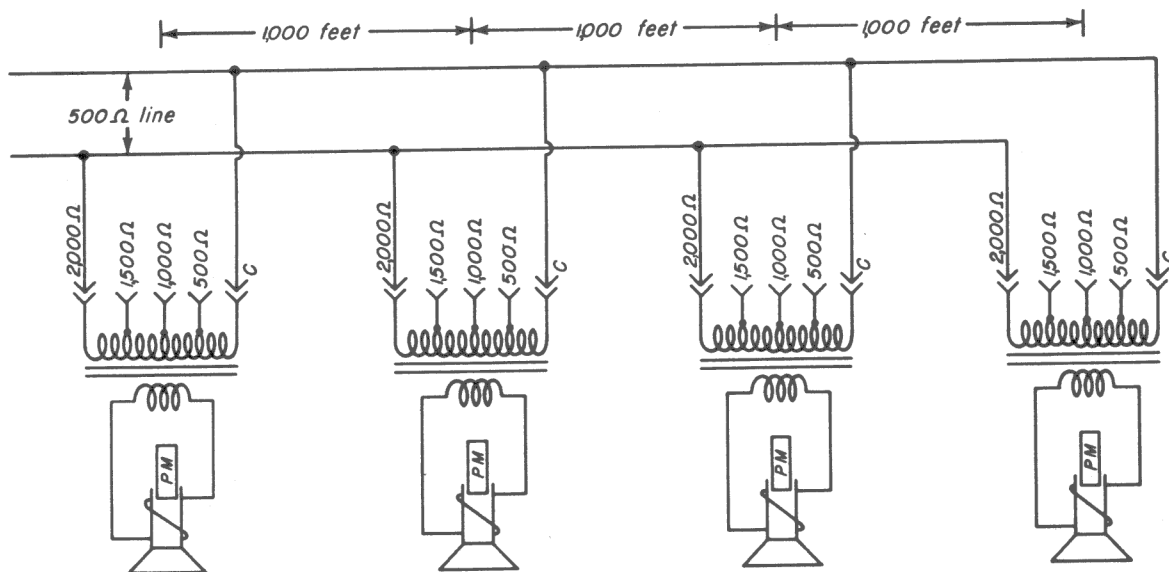


Fig. 27-19

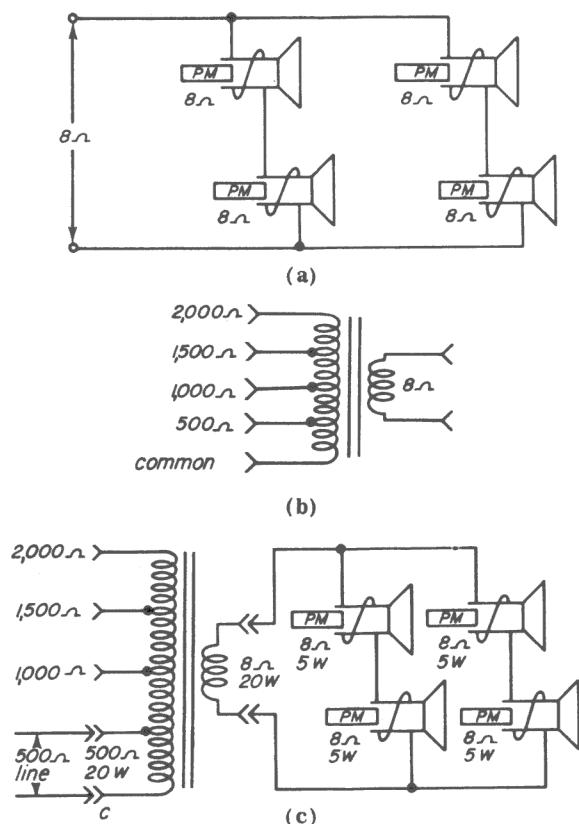


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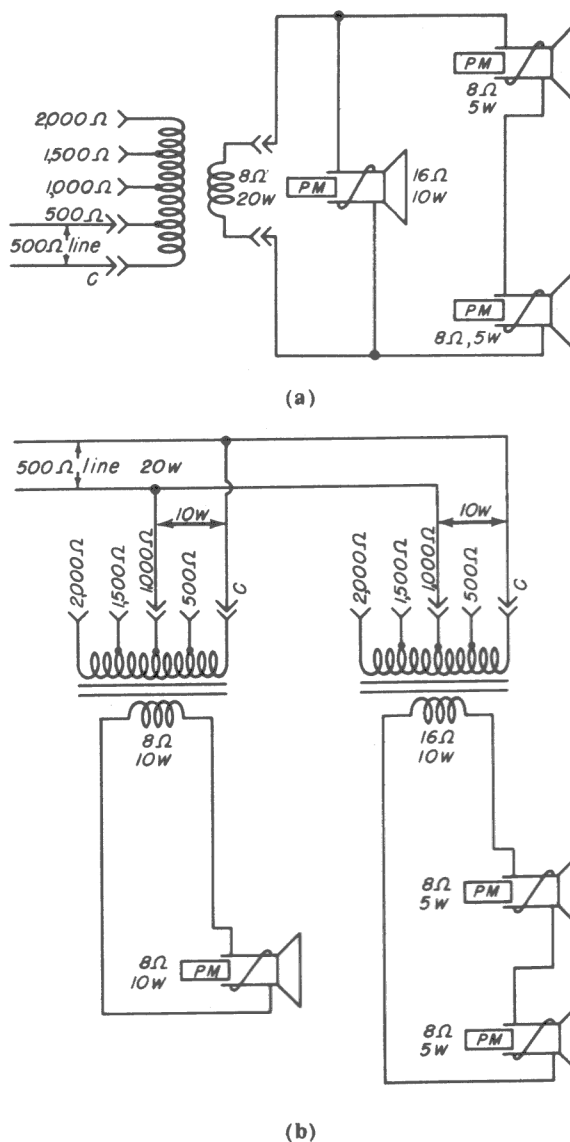


Fig. 27-18

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Constant-Voltage System. The constant-impedance system of matching loudspeakers to PA amplifiers has been in use for many years. But recently, a new system, called

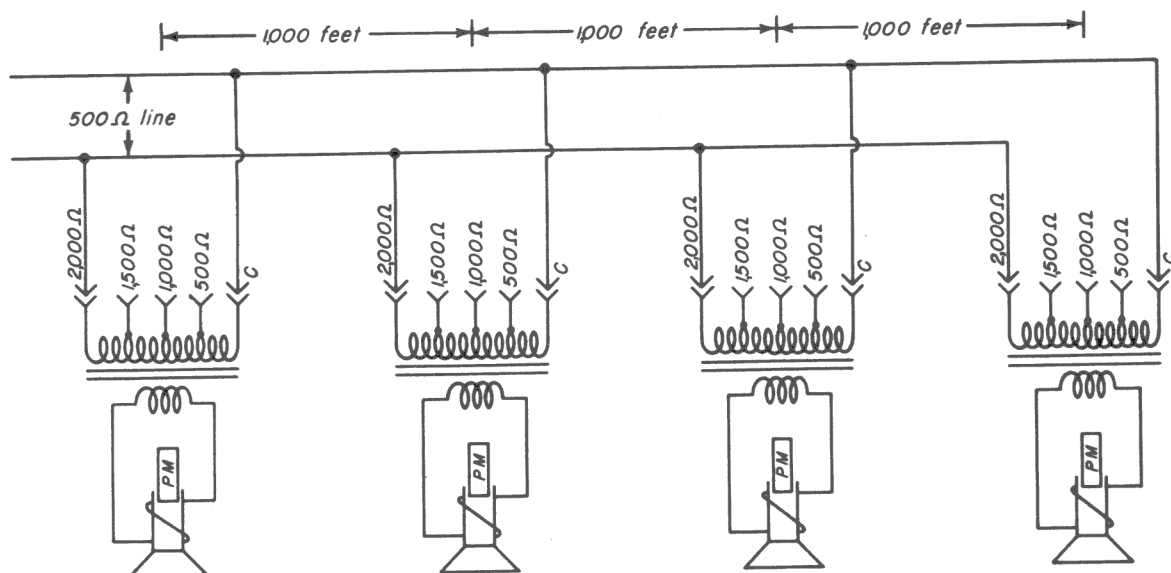


Fig. 27-19

the constant-voltage system, has come into use. Both systems can be classified as power-distribution systems. Let us compare the two systems and analyze the theory of each.

When the problem of distributing audio power to a group of loudspeakers presented itself originally—many years ago—it was natural for the constant-impedance system to develop as the most appropriate solution. Power amplifier tubes deliver maximum power output and minimum distortion to only one value of load impedance. So, for distributing the amplifier power output among several loudspeakers, it was natural to connect the speakers in a way to maintain the amplifier load impedance constant at the optimum value. This is what the constant-impedance system accomplishes. With the speakers connected in this way, the power delivered to the speaker load depends both upon the amplitude of the signal voltage which is put into the amplifier and upon the setting of the amplifier's volume control. In this respect, the system works just like a home radio receiver.

Power is affected by three electric quantities—voltage, current, and resistance (impedance). In the constant-impedance audio-power distribution system, impedance remains constant while voltage and current vary with the power delivered to the constant impedance load.

The electric power generation and distribution industry—the electric utilities—

were engaged in solving the technical problem of distributing a-c electric power at about the same time that the radio industry was engaged in solving its a-c audio power distribution problem. The electric utility industry maintains a constant voltage while allowing the impedance and current to vary. This is the constant-voltage system.

The difference between the constant-impedance system used in PA work and the constant-voltage system used for electric power distribution can be understood with the aid of Fig. 27-20.

Figure 27-20a shows an audio-power amplifier connected to a 500-ohm constant-impedance line. Let us list some basic facts concerning this setup.

1. The amplifier is not always set to deliver all the power it is able to deliver. The amount of power that it is set to deliver is determined by the setting of the volume control.
2. Although it is not set to deliver all the power it is capable of, it always delivers all the power it is set to deliver.
3. When you add or remove a speaker, you must do so in a way to maintain the impedance at 500 ohms. And when you do so, the power from the amplifier gets redistributed among the speakers, but the amplifier continues to deliver the same amount of power as before.

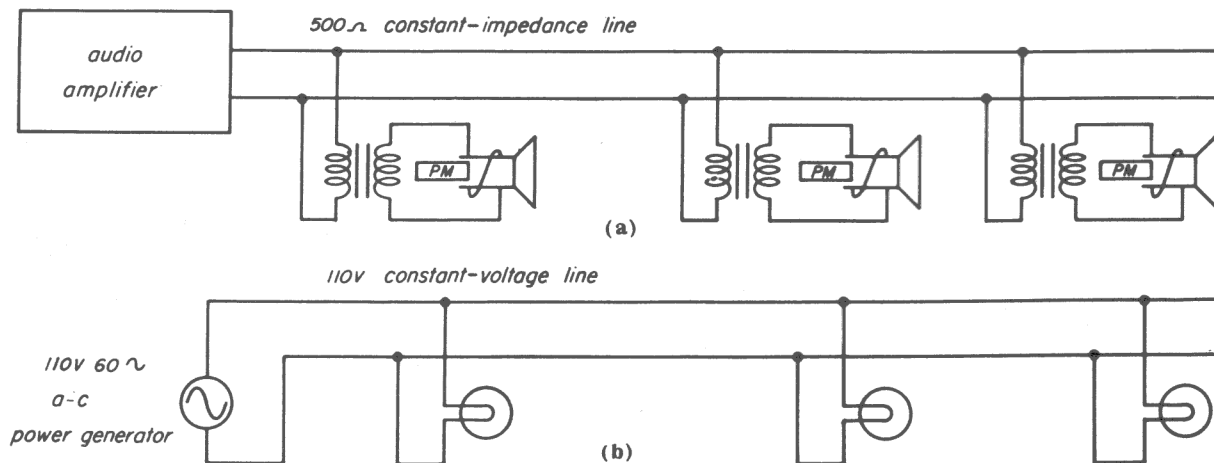


Fig. 27-20

4. The speakers are rated in terms of their impedance and the maximum power they can handle. If only one speaker is matched to the line, it gets all the power from the amplifier. If more than one speaker is matched to the line and you want to know how the power from the amplifier divides among them, you must compute this, taking the rated impedance of the speaker into consideration.

Now let us list a similar set of basic facts concerning the constant-voltage electric-power distribution system shown in Fig. 27-20b:

1. The generator is always set to deliver all the power it is able to deliver.

2. Although it is always set to deliver all the power it is capable of, it does not always deliver all the power it is set to deliver.

3. When you add or remove a lamp, you need not do so in a way to keep the impedance constant. And when you do so, the power in the remainder of the lamps does not change and does not get redistributed, but the generator delivers more or less power than before, depending upon whether the total impedance has been decreased by adding a lamp or increased by removing a lamp.

4. The lamps are rated in terms of the power they draw and the voltage level at which they are intended to operate (for example, a 60-watt, 110-volt bulb). If only one lamp is connected, it gets only as much power as it is designed to draw from the generator. If more than one lamp is connected, all the lamps draw their rated power up to the maximum power delivering capability of the generator. You need not compute the power distribution because you know it from the ratings of the lamps. (If you want to find the impedance, you have to compute it.)

The constant-voltage power distribution system used by the electric-power industry is handy, and it has been adapted to use in audio-power distribution systems. Let's briefly consider why it was not adapted

earlier and what now makes the adaptation possible. Then let's take up a couple of examples of using the constant-voltage audio power distribution system.

First, let's compare the electric industry's power generator to an audio power amplifier. Both deliver maximum power to only one optimum value of load impedance. Both deliver less power if a load impedance larger than the optimum value is connected, and both break down if a load impedance smaller than the optimum value is connected. However, unlike the generator, many audio amplifiers do not deliver a constant voltage. Also, while distortion of the a-c waveform is not a factor in electric-power generation and distribution, it is a factor in audio-power generation and distribution. Audio power amplifiers produce minimum distortion for only one value of load impedance. Because of the lack of constant-voltage audio-power amplifiers, and because of the distortion, the constant-voltage audio-power distribution system was not used in the past.

Today, audio-power amplifiers can be designed to deliver a constant voltage for a wider range of load-impedance values. Although distortion in these amplifiers is still greater at values of load impedance larger than the optimum, the distortion is not much greater, and is always within the limits tolerable in PA work. (Probably because of the distortion factor, the constant-voltage feature is not incorporated in amplifiers designed exclusively for high-fidelity applications not involving public address.) The design feature that makes these amplifiers possible is negative feedback. Here is a brief partial explanation of how negative feedback provides constant output voltage and low distortion. As the amplifier-load impedance is increased above the optimum value, the voltage across the load impedance increases, while the power in the load impedance decreases. But the feedback voltage is taken from across this impedance. So, when the impedance is increased and the voltage across it tends to increase, more voltage tends to be fed back, more of the input voltage tends to get cancelled, and, therefore, the output voltage tends to remain unchanged. This how feedback maintains a

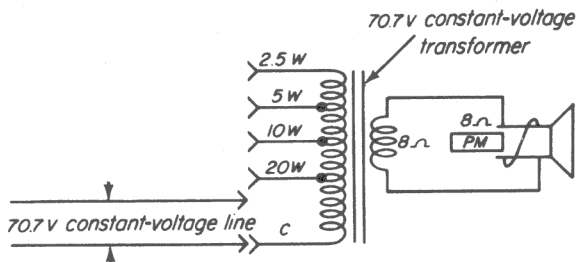


Fig. 27-21

constant output voltage with varying values of load impedance. Negative feedback keeps distortion low because the feedback signal voltage contains all the distortion that the amplifier is inclined to produce. By means of the feedback, the distortion is reintroduced to the amplifier input; since it is negative, it cancels the distortion being produced.

The standard value of constant voltage used in electric-lighting circuits is 110 volts. In adapting the constant-voltage distribution system to PA work, a voltage standard had to be decided upon. The Radio Manufacturers Association adopted a standard of 70.7 volts for all amplifiers delivering under 100 watts and 141 volts for all amplifiers delivering over 100 watts. Constant-voltage amplifiers deliver the standard voltage only when the volume control is set at maximum and the input-signal voltage is high enough to drive the power tubes to full output. As a result, a constant-voltage system should be planned and set up using the assumption of maximum input signal and

volume setting. This does not mean that all the power available from the amplifier will be delivered. The amount of power that is delivered depends upon the wattage rating of the speaker loads.

The system should be set up to deliver the most powerful sound that will ever be required. Once the system is set up, the volume can be turned down to reduce the sound from all the loudspeakers at once or the tap on the transformer associated with each speaker can be changed to reduce the power drawn by each speaker.

The speaker loads are rated in wattage drawn and the voltage level at which they are intended to operate in the constant-voltage system, rather than in terms of impedance and maximum power handling ability as in the constant-impedance system. This corresponds to the wattage/voltage ratings of electric-light bulbs. However, unlike the bulbs, the speakers are equipped with a transformer. Changing the transformer tap that is joined to the 70.7-volt line changes the power drawn by the speaker. The setup is shown schematically in Fig. 27-21. Some speakers are fitted with the transformers. If a speaker does not already have such a transformer, separate constant-voltage transformers may be purchased. A constant-voltage transformer can be connected to any speaker with the correct voice-coil impedance. Changing the tap changes

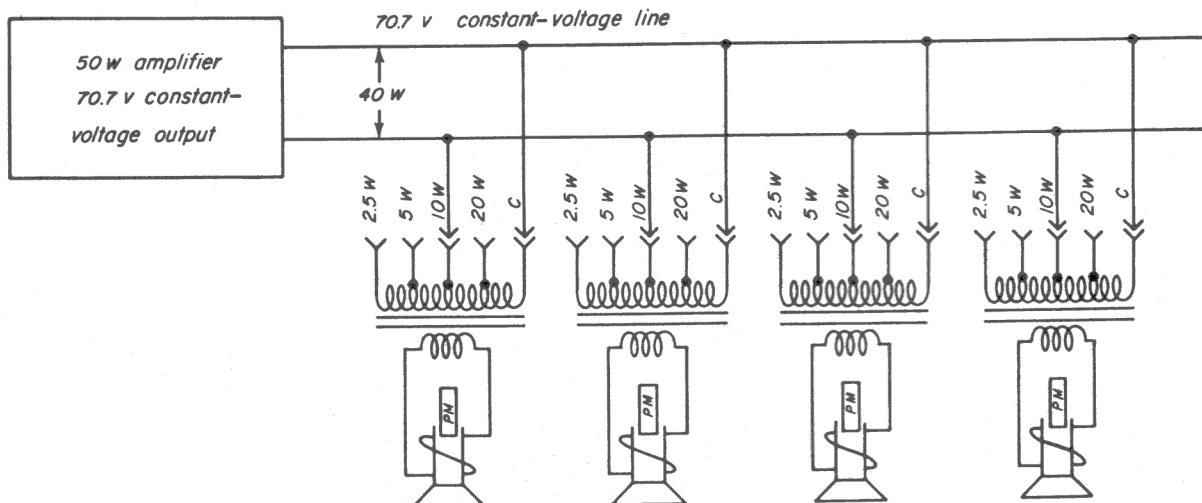


Fig. 27-22

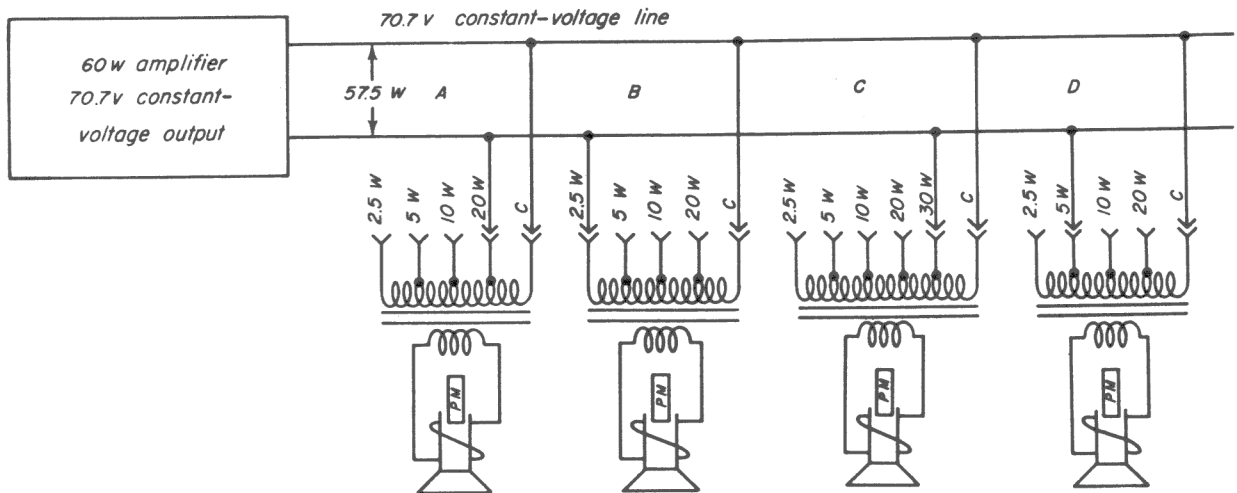


Fig. 27-23

the impedance presented to the line but it is not necessary to match the speaker impedance to the line impedance. In fact, both impedances may be unknown. Nevertheless, the standard voltages (70.7 volts and 141 volts) were chosen to result in a line impedance sufficiently high for long-distance power transmission.

To take an example, suppose we want to connect four speakers to the end of a 70.7-volt line. We have a 50-watt amplifier and four speakers equipped with taps rated at 2.5 watts, 5 watts, 10 watts, and 20 watts. We want each speaker to draw 10 watts at most. All we need do is set the volume to maximum and connect the 10-watt tap of each speaker to the line, as shown in Fig. 27-22. Each speaker will draw 10 watts, and the

amplifier will deliver only 40 watts of its 50-watt maximum power output even though the volume control is set at maximum.

Suppose we want the speakers to be spaced at intervals along the line at positions A, B, C, and D in Fig. 27-23. At position A, we want 20 watts; at position B, we want 2.5 watts; at position C, we want 30 watts; and at position D, we want 5 watts. The total power to be drawn adds up to 57.5 watts, and our amplifier is rated at only 50 watts maximum. We will have to get another amplifier—one rated at 60 watts maximum. Also, we want 30 watts at position C and our speakers have taps up to only 20 watts. We need one new speaker with a 30-watt tap. With the new speaker and the new amplifier, we can set things up as in Fig. 27-23.